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SPEECH OPTIMIZATION AT 9600 BITS/SECOND

DCA 100-78-C-0064

VOLUME 1

SOFTWARE SIMULATION

SUBMITTED TO
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SEPTEMBER 30, 1980

Sylvania Systems Group
Communication Systems Division
GTE Products Corporation
77 A Street
Needham Heights, Mass. 02194 U.S.A.
Area Code 617 449-2000
TELEX: 92-2497

GTE | Systems

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(10) A.J./Goldberg L./Cosell S./Kwon L./Bergeron R./Cheung

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20. ABSTRACT (Continue on reverse side if necessary and identify by block number)

This report describes the design and development of a real-time adaptive transform coder that transmits high-quality speech over a 9600 bps channel with bit-error rates of up to 1% without significant loss of speech fidelity. The report presents the results of our FORTRAN simulations on the adaptive transform coder which maximized the quality of the transmitted speech. Important aspects of the ATC algorithm which are optimized
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20. Abstract (cont'd)

were specification and transmission of the side-band information, accuracy of the pitch and voicing decisions, and error-protection of the important transmission parameters. Also included is the system design, detailed documentation, and program listings of the MAP-300 real-time implementation of the optimized ATC speech coder. Finally, the report includes a description of analog equipment GTE built to interface the MAP-300 to telephone handsets and tape recorders and a description of digital circuits (RS 423 compatible) to interface the MAP-300 to a modem.

This report is bound in two volumes. Volume I contains a description of the ATC system and the results of the FORTRAN simulations. Volume II contains all the information on the real-time system including documentation for implementing the ATC system on the MAP, listing of the MAP software, and documentation for the hardware built by GTE.

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Chapter I

Summary of Program

1.1 Introduction

Under the 9600 BPS Speech Optimization Study, GTE Sylvania simulated and implemented a full duplex Adaptive Transform Coder (ATC) speech digitization algorithm. The simulations were performed using FORTRAN computer programs while the implementation used a CSP,Inc. Map -300 floating point array processor with digital and audio input/ouput circuitry designed by GTE.

This study and implementation effort has resulted in a number of significant accomplishments in developing speech digitization algorithms. The most important of these include:

a. The demonstration via FORTRAN simulation that ATC at 9600 bps can produce good quality speech having a Signal-to-Noise ratio (S/N) of about 17 dB.

b. Establishment of a benchmark speech processing technique at 9600 b/s which indicates that high quality speech is possible at this data rate.

c. The development of error coding techniques which will permit ATC to function at a bit error rate (BER) of 10^{-2} with little reduction in S/N using (63,45) BCH codes.

d. The design and implementation of analog audio circuitry to permit speech to be input to and output from the CSP, Inc. MAP processor from microphones and tape recorders.

e. The design and implementation of a digital transmission interface (RS 423 compatible) to the CSP, Inc. MAP processor so that data can be sent to a modem.

f. Implementation of a real-time full duplex ATC speech digitizer on the CSP, Inc. MAP processor whose block diagram is shown in Figure 1.1-1. This digitizer performs its processing with floating point arithmetic and, does not compromise numerical accuracy.

g. Real-time demonstration of ATC in the presence of 10^{-2} channel error rate without significant performance degradation.

The voice quality produced by the ATC simulation is the best of any technique operating at 9600 b/s now known to GTE. The technique, whose specifications are shown in Table 1.2-1, is numerically complex requiring the complete processing capability of the CSP, Inc. MAP-300 floating point processor. Thus, for ATC to be practical, either higher speed hardware must be built or the technique must be simplified.

The investigation and developments leading to the real-time ATC system proceeded in three phases. During the first phase the ATC algorithm originally proposed by Zelinski and Noll[1,2] was investigated and the modifications proposed by Crochiere and Tribolet[3,4] were incorporated to improve voice quality. Numerous FORTRAN simulations were conducted to optimize performance with respect to data rate, channel error performance and robustness to speaker and room noise. At the end of the first phase which lasted about 4 months the ATC algorithm was frozen and the real-time implementation begun.

Concurrent with the first phase was the design and fabrication of the digital and analog I/O interfaces to the CSP, Inc. MAP-300 processor. In addition to building our own units, GTE Sylvania, under separate subcontracts with BBN and Notre Dame University, built two additional units for incorporation with their MAP-300 speech processing systems.

SPEECH PROCESSING SYSTEM COMPRISED OF MAP HARDWARE AND SPEECH PROCESSOR INTERFACE

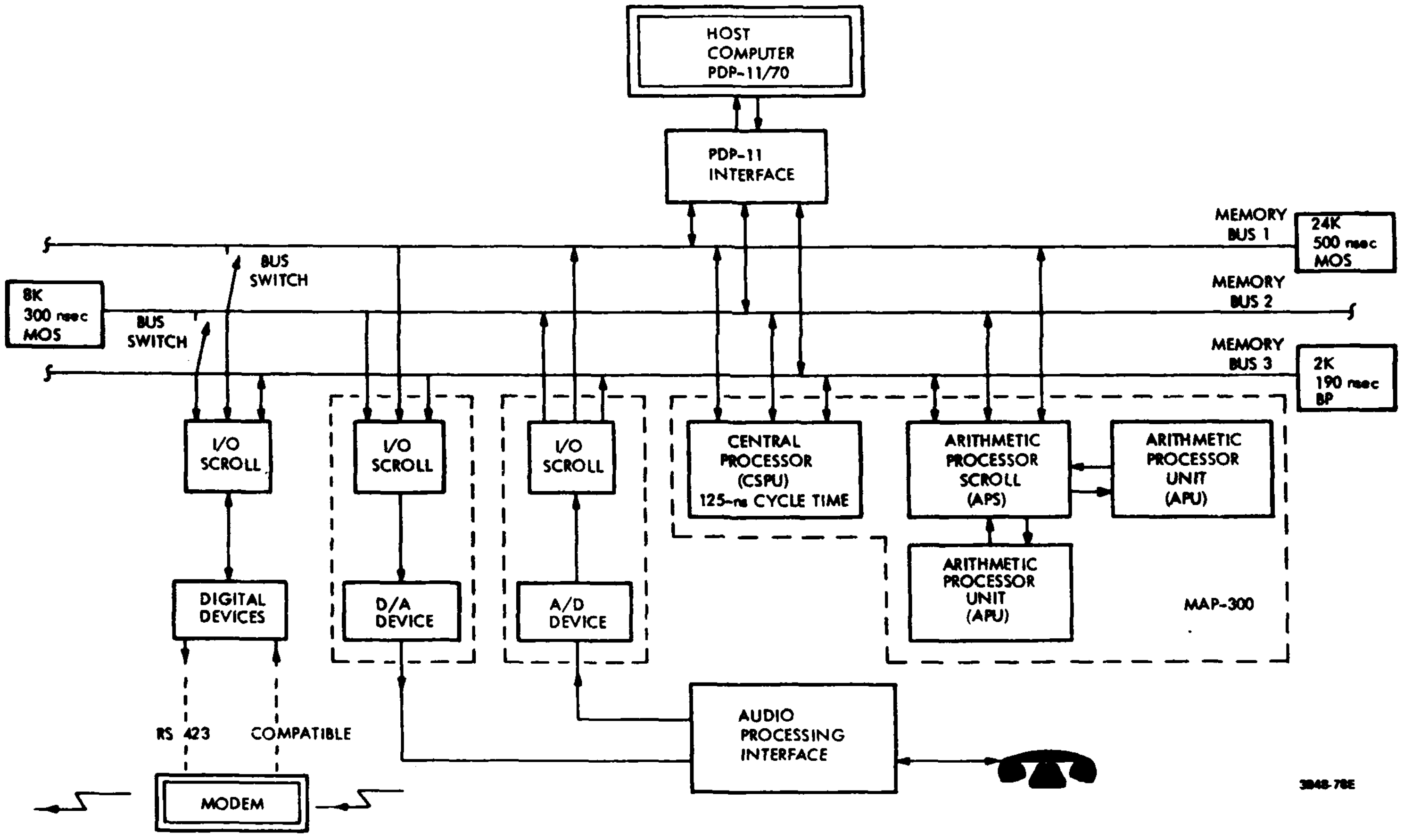


FIGURE 1.1-1

PARAMETER	SPECIFICATION
Input Bandwidth	0-3200 Hz
Sampling Rate	6400 Hz
Frame Rate	26.016/sec.
Number of Samples/Frame	246
Number of Samples Overlapped/Frame	10
Bits/Frame	369
Pitch	6 if voiced
	0 if unvoiced
Pitch Gain	2 if voiced
	0 if unvoiced
Voiced/Unvoiced	1
RMS Energy	5
DC BIAS	5
PARCOR 1	5
PARCOR 2	5
PARCOR 3	4
PARCOR 4	4
PARCOR 5	3
PARCOR 6	3
PARCOR 7	2
PARCOR 8	2
Parity Bits (Error Correction)	54
SYNC	1
DCT Coefficients	267 voiced
	275 unvoiced
Number of Error Control Blocks/Frame	3
Error Control Technique	(63,45) BCH

TABLE 1.1-1: OPTIMIZED ATC S[illegible]M SPECIFICATION

The third phase, the real-time implementation, began in February 1979 and continued until August 1980. During this time, test programs for the analog and digital I/O were developed and numerous software and hardware problems with the MAP-300 were resolved. Finally, in the summer of 1979, the first working modules of the ATC digitizer were operational on the MAP-300, and it was at this time that the scope of the software development project became apparent. The MAP-300, for all its speed was barely adequate to perform ATC with error control in a full duplex mode. Consequently, from August 1979 to the delivery of the ATC system a year later, considerable effort was placed on writing efficient MAP-300 real-time software.

The final ATC system, as delivered to DCA, indicates that a full duplex ATC speech processing system can operate on the MAP-300 processor in real-time.

Future speech digitization development at 9600 cannot ignore the ATC algorithm because even though the technique is complex, it shows that good quality speech is possible at this data rate. Thus, the ATC technique developed under this contract will serve as a benchmark or standard to compare all new 9600 b/s speech digitization algorithms.

This report is written in two volumes. Volume 1 contains documentation on the ATC simulations while Volume 2 contains documentation on the real-time software and hardware I/O circuitry.

Chapter 2

Simulation of the ATC Algorithm

2.1 Introduction

Adaptive Transform Coding (ATC) was originally proposed by Zelinsky and Noll [1,2] and represents an efficient block-coding technique for speech digitization in the 8.0 to 16 K b/s range. Early simulations of the ATC algorithm at GTE Sylvania indicated that this technique was capable of producing better speech quality than any other technique at 9.6 kbps known to the company at the time. When DCA requested the study of new techniques at 9.6 kbps GTE Sylvania responded with the ATC algorithms as originally proposed by Zelinsky and Noll. Later articles by Tribolet and Crochiere[3, 4] however, indicated that further improvements were possible in the algorithm, and after contract award, GTE decided, based on simulations conducted under its IR&D program, to develop this algorithm even though it was about 50% more complex than the original Zelinsky and Noll design.

In this Chapter we first discuss the theory of ATC operation. Then we discuss the simulation and optimization of this system and the need for error protection and correction for some critical transmission parameters. Finally we discuss the results of the simulation with and without error protective coding in the presence of channel errors as high as one error in 100 bits. (A BER of 10^{-2}.)

2.2 Basic Principles of ATC Operation

In its basic form, ATC consists of sending the largest cosine transform coefficients of a segment of data with each coefficient quantized according to an algorithm that gives the larger coefficients more bits than the smaller coefficients. This ATC algorithm departs from earlier algorithms that not only had to send the amplitudes of the coefficients, but also had to send considerable information about which coefficients were quantized and how many bits were associated with each. This extra information could consume as much data capacity as the coefficient amplitudes themselves. Attempts at sending only specific coefficients or the use of a fixed-bit assignment generally reduced voice quality by creating waveform discontinuities at the frame boundaries and by spectrally distorting the signal between boundaries.

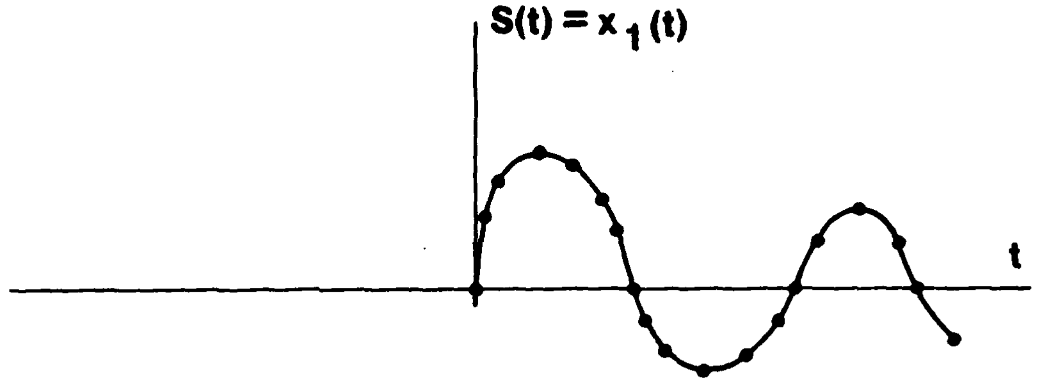
In ATC, however, information about which amplitude is sent and how many bits are allocated to each is contained in the basis spectrum, which requires from 1200 to 2400 b/s. This basis spectrum generally is information about the envelope of the transform coefficients being quantized. Its calculation can be performed by the smoothing of transform coefficients or by separate estimates involving least-square analysis.

To understand ATC, consider a sampled waveform segment shown in Figure 2.3-1(a). If this waveform is multiplied by 1/2, delayed by half the sampling interval T, and reflected about t=0, it yields $X_2(t)$ whose Fourier transform is given by:

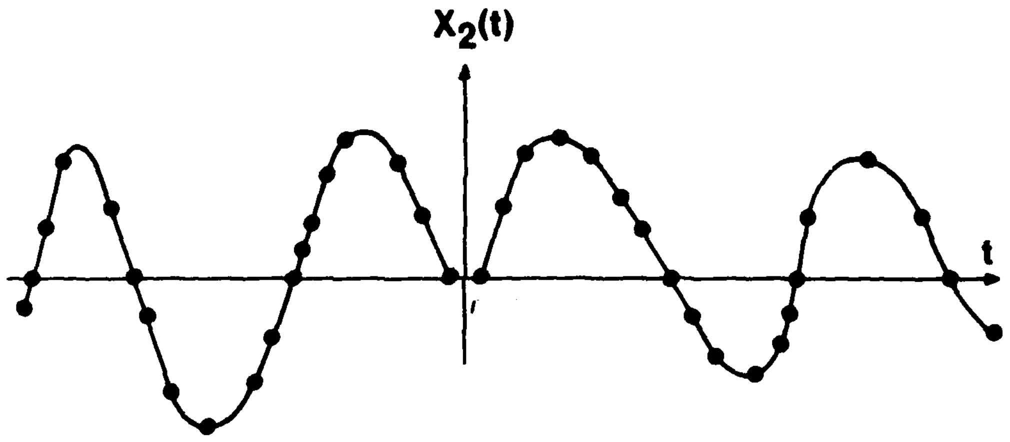
$$X_2(f) = \sum_{n=-(N-1)}^{N-1} x_2(nT) \exp\left(-j2\pi f(n+1/2)T\right) \qquad (2.2\text{-}1)$$

If we sample the Fourier transform of $X_2(f)$ at frequencies $\frac{\pi}{2NT}$, the discrete Fourier transform (DFT) becomes

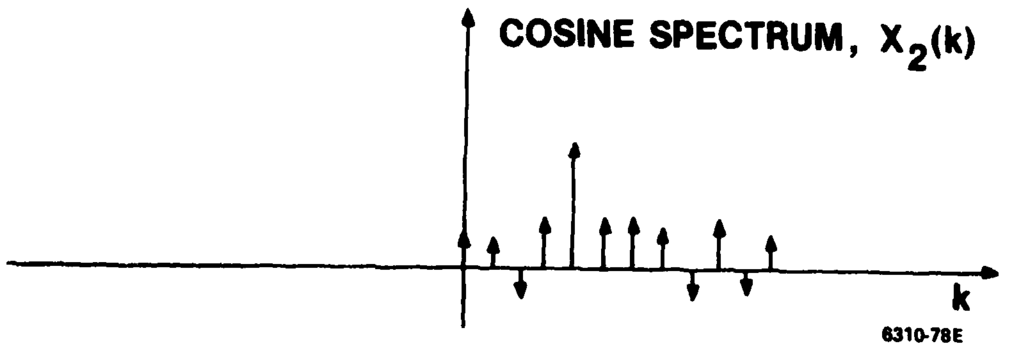
$$X_2(\frac{m}{2NT}) = X_2(m) = \sum_{n=-(N-1)}^{N-1} X_2(nT) \exp(-j\frac{\pi m}{N}(n+1/2)) \qquad (2.2\text{-}2)$$



(a) Original Waveform



(b) Reflected Waveform



(c) DFT Output

Figure 2.2-1: Discrete Cosine Transform Operation

Using symmetry properties of $X_2(nT)$, $X_2(m)$ shown in Figure 2.2-1(b) is real only and is given by

$$X_2(m) = \sum_{n=0}^{N-1} X_1(nT) \cos\left(\frac{\pi m}{2N}(2n+1)\right) \quad 0 \le m \le N-1 \qquad (2.2\text{-}3)$$

Equation (2.2-3) is the cosine transform. This derivation shows that the Fast Fourier Transform (FFT) can be used to implement the cosine transform by delaying and reflecting the original waveform and then taking the FFT on a waveform twice as long as the original.

The most expensive implementation costs with the ATC algorithm are associated with the Discrete Cosine Transform (DCT) and Discrete Fourier Transform (DFT). Although the DCT cannot be employed directly, methods elaborated by Ahmed et al[6] and Cooley et al[7] use the DFT to compute the desired transform. Our FORTRAN simulations used the Cooley method for DCT calculation and a special FFT algorithm to lower simulation costs.

After calculation of the DCT coefficients, the basis spectrum (envelope of the cosine transform) can be estimated by making all the cosine transform coefficients positive and smoothing between peaks to efficiently send the envelope. We can quantize the amplitudes of every mth (m is typically 8) envelope sample and send those as the coefficients of the basis spectrum.

However, this original ATC algorithm, as proposed by Zelinski and Noll, suffers from a "burbling" characteristic at lower data rates. To reduce this distortion, Tribolet uses side transmission of pitch and spectral parameters obtained by Linear Predictive Coding (LPC)[5] analysis. The side transmission of the LPC and pitch parameters does in fact remove the "burbling" sound and improve the overall signal-to-noise ratio. Figure 2.2-2 describes

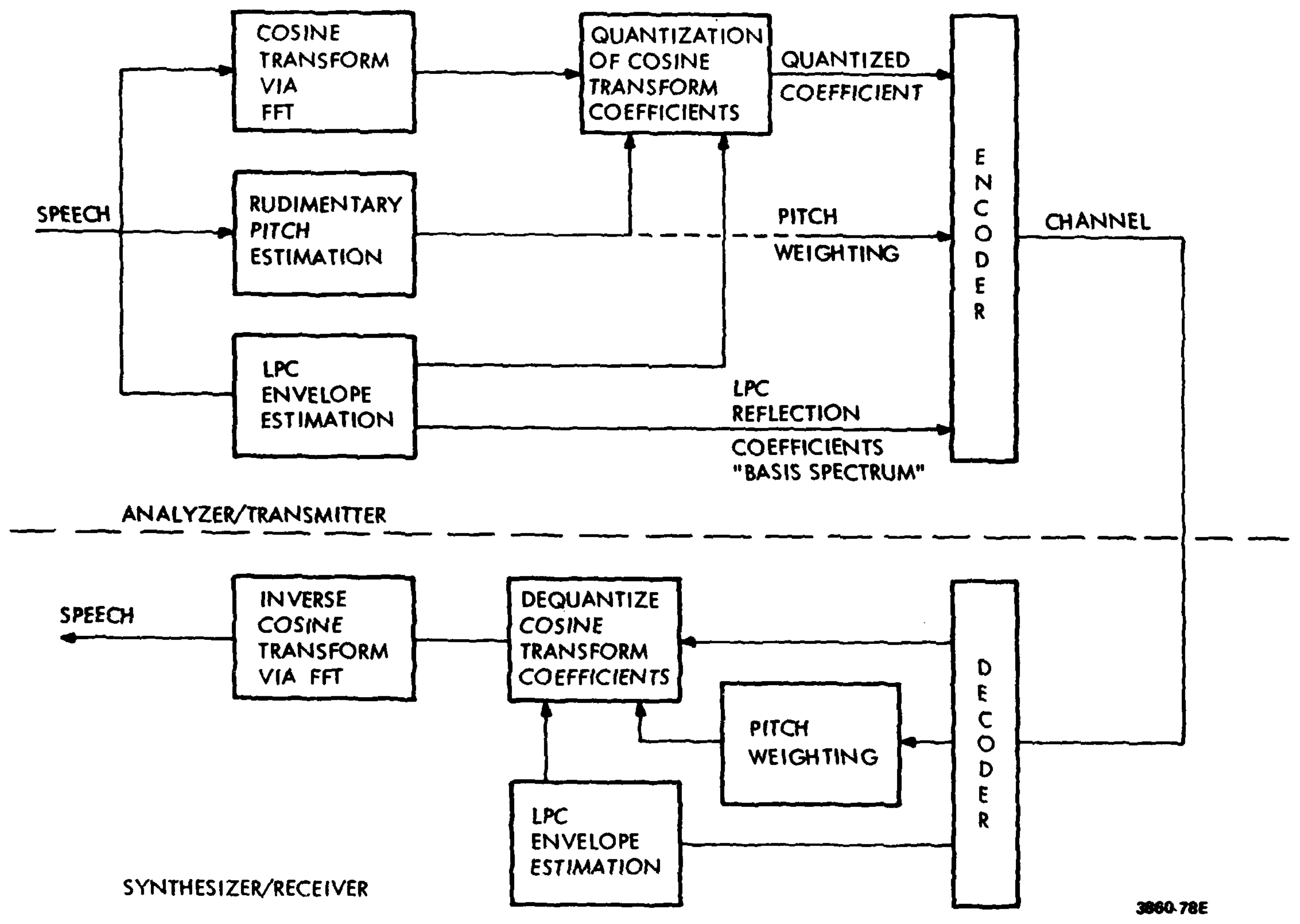


Figure 2.2-2: Adaptive Transform Coder

the operation of this ATC digitizer.

The innovative solution to the basis spectrum calculation is formed from a least-square analysis of $x_2(t)$, that is, finding those predictor coefficients which minimize.

$$E = \sum_{n=0}^{N-1} \left[X_2(nT) - \sum_{i=1}^{P} a_i X_2 \left[(n-i)T\right] \right]^2 \qquad (2.2\text{-}4)$$

These predictor coefficients, or alternately reflection coefficients, carry information about the envelope since:

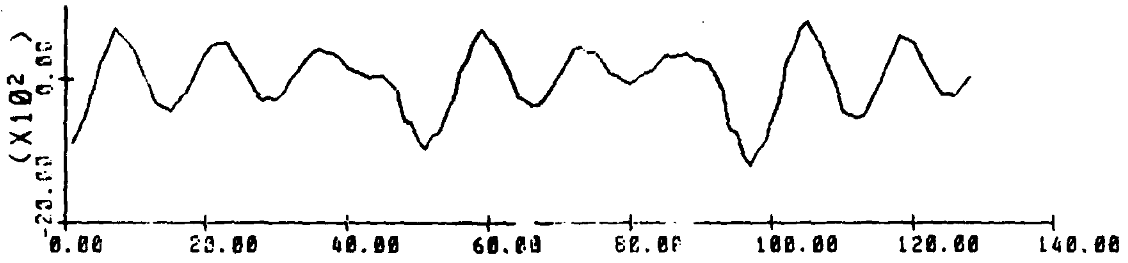
$$Y(f) = FFT(a_i) \qquad (2.2\text{-}5)$$

and the envelope is then $Y^{-1}(f)$.

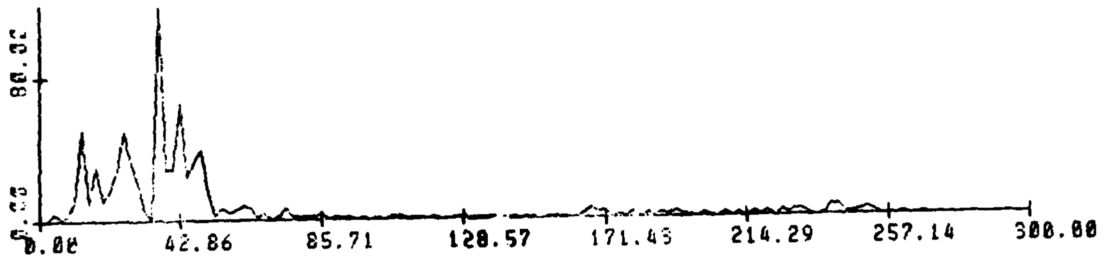
In addition to linear predictive modeling of the ATC spectrum, the Tribolet approach uses a pitch excitation source. This accounts for the fine structure in the short-time spectrum, which is consistent with the known mechanisms of speech production. This scheme forces the assignment of transform bits to many pitch striations that otherwise would not be transmitted at all.

With reference to Figure 2.2-3, the ATC analysis is described as follows:

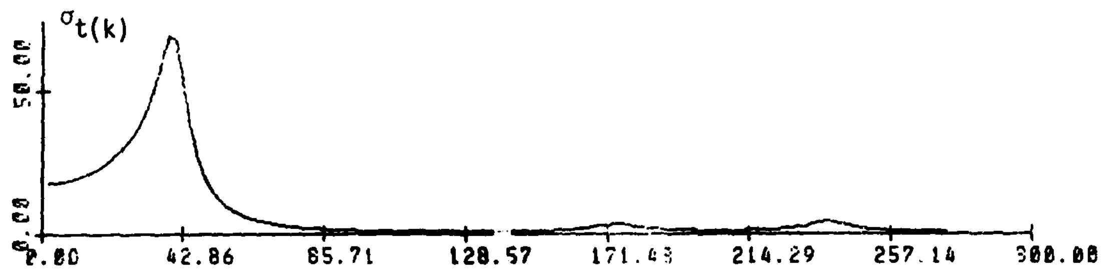
1. The input speech (Figure 2.2-3(a)) is Fourier transformed to yield a DCT spectrum (Figure 2.2-3(b)). This spectrum is squared, windowed, and inverse Fourier transformed to yeild an autocorrelation function (i.e., pseudo-ACF) of the reflected speech waveform. The first P+1 values of this function are used to define a correlation matrix in the usual normal equation formulation sense. The solution of these equations (i.e., Levinson recursion) yields a prediction filter of



(a) Input Speech Samples



(b) DCT of Input Speech(X10^1)



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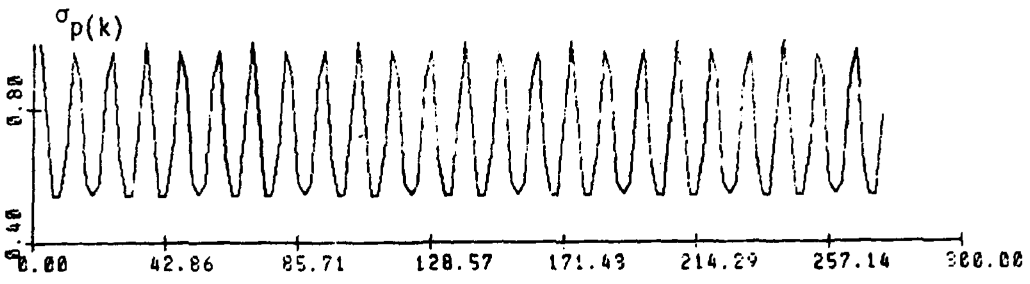
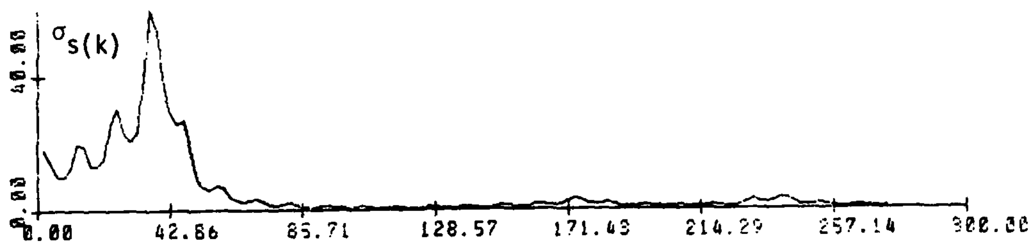
(c) LPC Spectrum(X10^1)

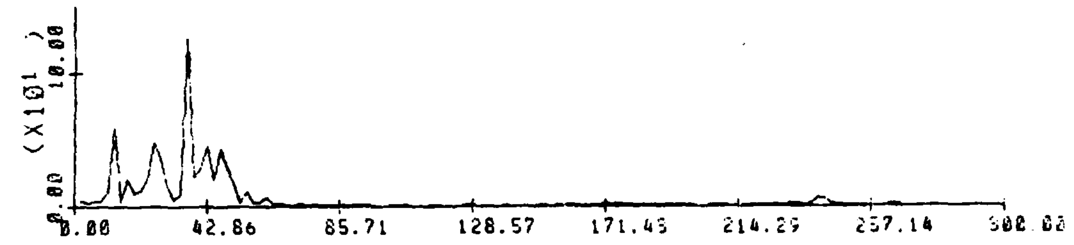
Figure 2.2-3: Graphical Description of Vocoder Strategy for ATC



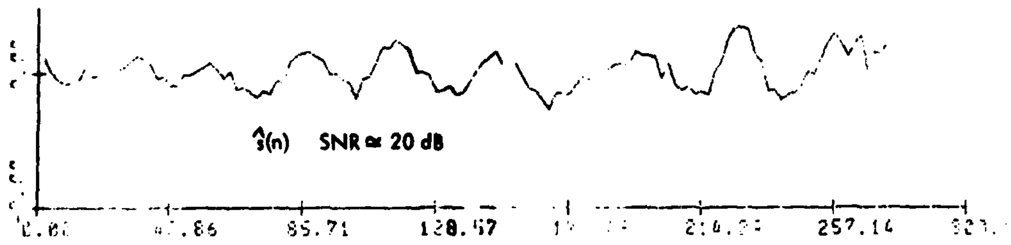
(d) Pitch-Weighting Spectrum(X10^1)



(e) Basis Spectrum(X10^1)



(f) Quantized DCT(X10^1)



(g) Error Waveform-Original-Processed(X10^1)

800-79E

Figure 2.2-3: Graphical Description of Vocoder Strategy for ATC (Cont.)

order P. The inverse spectrum of this filter yields a smoothed estimate of the DCT (Figure 2.2-3(c)) spectrum levels to be used in the adaptation of the quantizers.

2. A rudimentary estimate of the pitch value, M, is found in the pseudo-ACF after the second zero crossing beyond the P+1 ACF value. A corresponding gain factor, G, is also computed as the ratio of ACF(M)/ACF(0). With these two parameters, a pitch pattern is generated in the frequency domain (Figure 2.2-3(d)) and applied congruently with the LPC spectrum. This combination, yielding a linear prediction spectral fit to the DCT of the input speech, is called the basis spectrum (Figure 2.2-3(e)).

3. The computation to determine the number of bits to allocate for each transform then proceeds as follows:

Let σ_i be the amplitude of the ith term of the envelope of the basis spectrum. The B_i, the number of bits allocated to the ith cosine transform coefficient, is given by:

$$B_i = \left[B_f/N - (1/2N) \sum_{j=1}^{N} \log_2 \sigma_j^2 \right] + 1/2 \log_2 \sigma_i^2 \qquad (2.2\text{-}6)$$

where

B_f = the total number of bits allocated to send the cosine transform coefficients per frame

N = the total number of cosine transform coefficients calculated per frame.

Note that the term in brackets is calculated once per frame. Fairly simple algorithms ensure that B_i is an integer value and that the sum of the integer B_i adds to B_f.

The cosine transform coefficients approximate a Gaussian probability density function. Optimum Gaussian quantizers derived by Max[20] can be used to encode each transform coefficient with B_i bits. Since many of the B_i's will be zero, only larger coefficients are sent. However, GTE Sylvania's experience with the 9600-b/s ATC has shown that optimal quantizers can be developed that more closely match the transform distribution.

4. The receiver uses the basis spectrum information (LPC, M, G) to regenerate the DCT envelope, to generate the bit allocation using Equation (2.2-6), to decode the cosine transform coefficients (Figure 2.2-3(f)), and then to take the inverse cosine transform using the FFT. Frame boundary problems exist at all data rates since quantization of the transform coefficients causes the regenerated waveform to be slightly different than the original. By overlapping the frames slightly and by interpolating across the frame boundaries, these discontinuities can be smoothed.

The overall quality of this approach can be estimated from Figure 2.2-3(g), which shows the error waveform defined as:

$$e(n) = s(n) - \hat{s}(n) \tag{2.2-7}$$

The received waveform, $\hat{s}(n)$, has a high signal-to-noise ratio ($\sim$17 dB) for some speakers, even for erroneous pitch estimations made in the analyzer. In fact, GTE Sylvania has demonstrated through audio tapes that an eighth-order LPC predictor ($P = 8$), coupled with the rudimentary pitch extractor (and no voiced/unvoiced logic), yields consistently high-quality speech.

2.3 Optimization and Modification of the ATC System

The adaptive transform coding scheme shown in Figure 2.2-2 produces high quality synthesized speech above 9600 bps. In this scheme, the quality in objective signal-to-noise ratio and in subjective perceptual effects degrades by lowering the transmission data rate.

There are several sources which reduce the voice quality but solutions for most of these problems exist. The first one is the quantization noise caused by coarse quantization of the DCT coefficients at low data rates. This problem can be minimized by developing optimal quantizers from the distribution of actual DCT coefficients. The second and most severe degradation source is the reduction in bandwidth at low data rates. This effective lowpass filtering stems from the fact that only large DCT coefficients are being coded because there are not sufficient bits to send the smaller coefficients. The effects of lowpass filtering can be removed by the addition of random noise to the low energy frequency band. The third source of degradation are waveform discontinuities at the frame boundaries since the DCT coefficients are coarsely quantized and some low valued coefficients are not transmitted. These effects may be reduced by overlapping the frames slightly and by interpolating across the frame boundaries.

There are other areas for improvement in the ATC scheme. The first one is the trade-off of bits allocated to DCT coefficients and to side information within given transmission data rate. Another is the method for quantizing the side information. It can be shown that closer estimation (in the mean square error sense) of the basis function to actual DCT coefficients provides better performance of the ATC coder. In fact, the extreme case, where the basis spectrum equals the actual spectrum, quantization of the DCT can be precise, eliminating lowpass effects or boundary problems as long as the sign bits of DCT coefficients are provided because DCT is a unitary transform.

In the following sections, the modified ATC system will be described. Also, those areas requiring further developments will be described and the possible improvements will be discussed.

2.3.1 Description of the Modified ATC System

The block diagram of the ATC analyzer and synthesizer are shown in Figure 2.3-1 and Figure 2.3-2, respectively. In this scheme, the input speech is buffered into blocks of data {v(n)} which consist of a frame. This frame of speech data is overlapped slightly (about 10 samples) in order to reduce the frame boundary problems. The mean and variance of the input speech signal are calculated for the transformation of zero mean and unit variance. This mean and variance are quantized and sent to the receiver for the renormalization of the synthesized speech. The Discrete Cosine Transform is calculated on the zero mean and unit variance input. The DCT coefficients are then adaptively quantized to form mainband information and transmitted to the receiver. At the receiver, they are decoded and inverse transformed to reproduce the zero mean and unit variance speech signal. This signal is renormalized by the mean and variance to produce the synthesized speech.

In the meantime, the mean and variance of the signal are decoded and dequantized to renormalize the inverse transformed signal. In order to reduce the effects of signal discontinuities at the frame boundaries, the overlapped signals are interpolated.

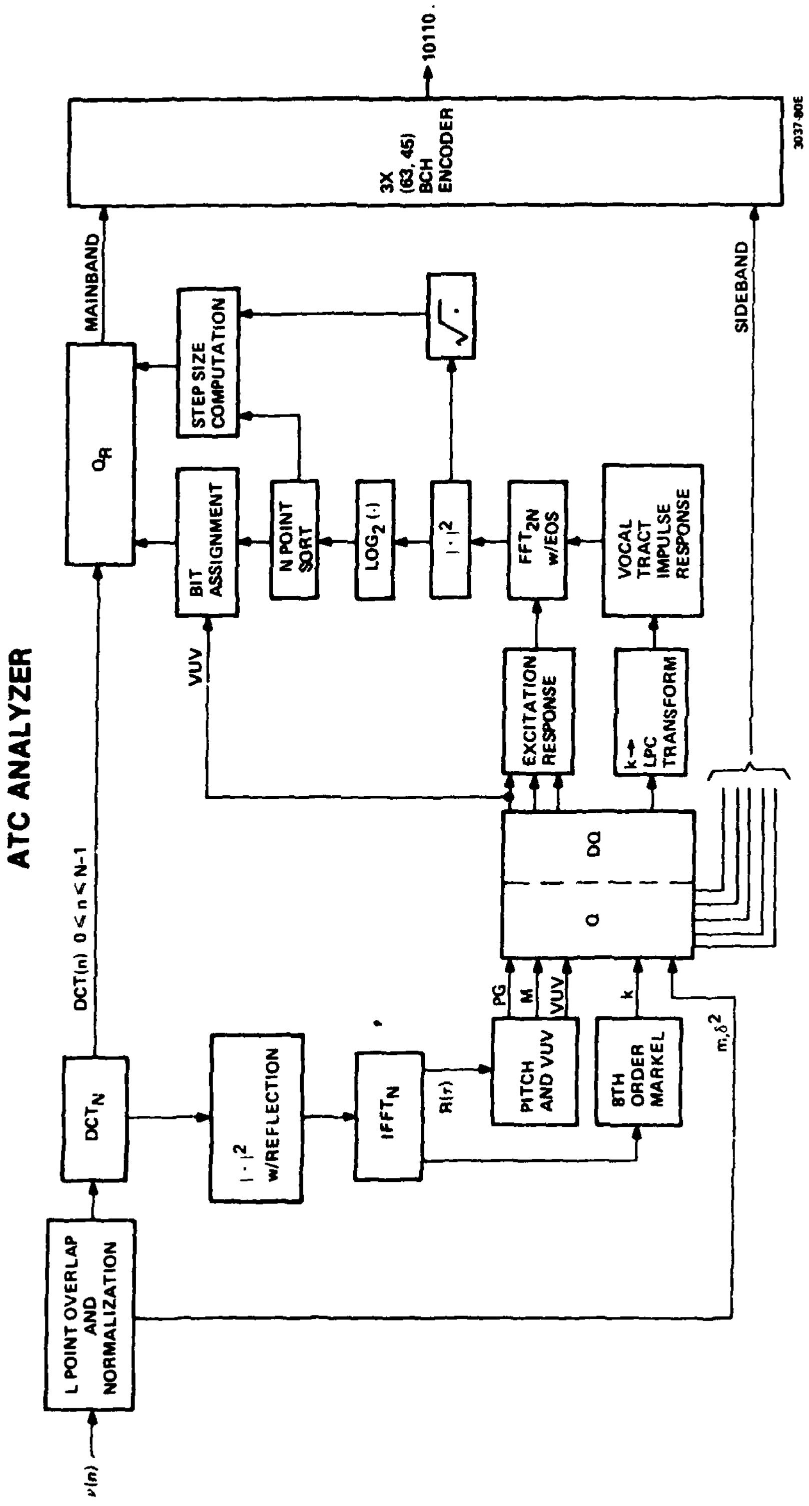


FIGURE 2.3-1 BLOCK DIAGRAM OF THE ATC ANALYZER

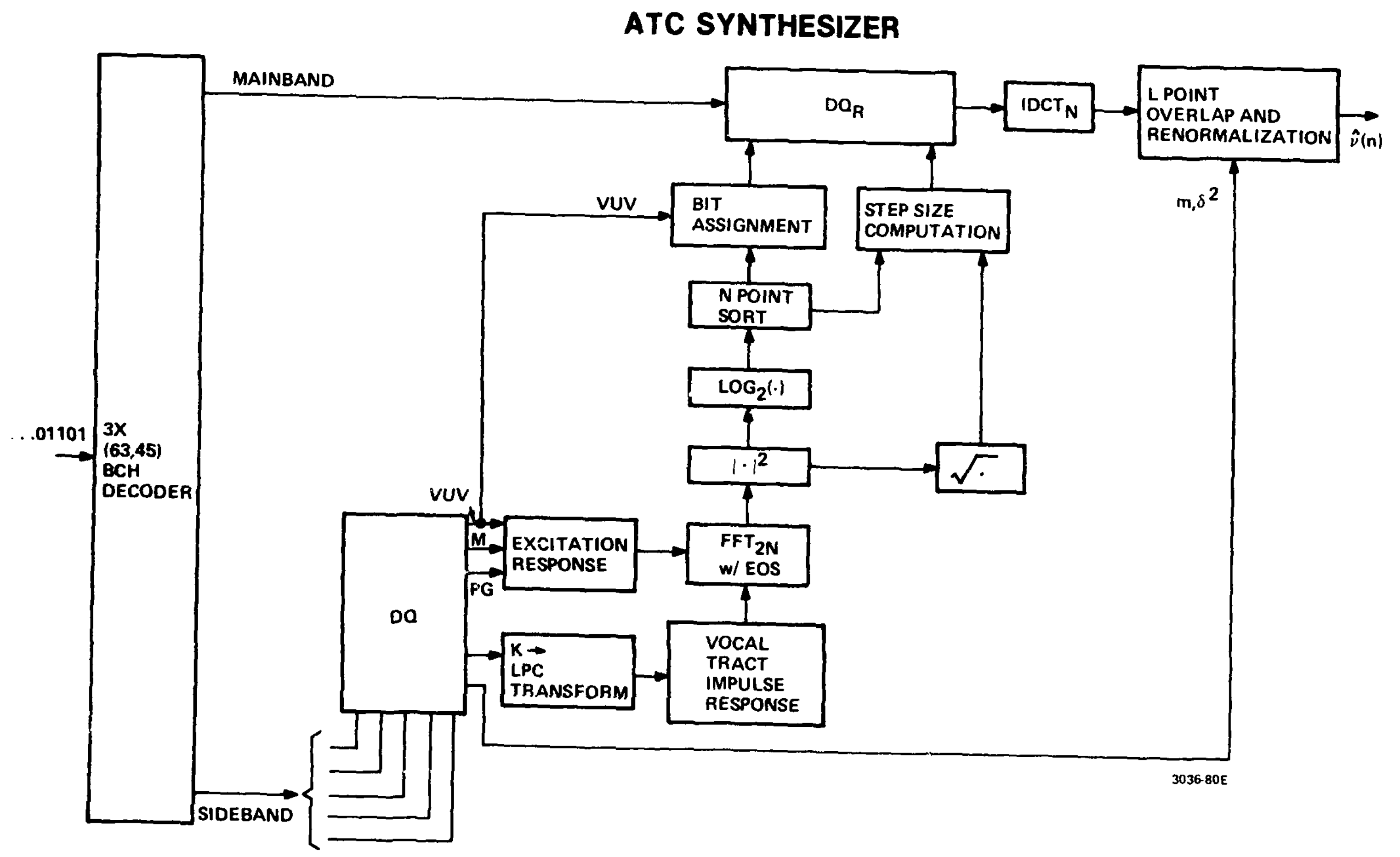


FIGURE 2.3-2 BLOCK DIAGRAM OF THE ATC SYNTHESIZER

The adaptive quantizations and dequantizations of this scheme are based on the sideband information which the basis spectrum will be computed from. The bit assignments and step size computation will be determined by the optimum bit assignments rule from the basis spectrum. The sideband information includes the pitch gain (PG), pitch number (M), voiced/unvoiced decision, and the 8 PARCOR coefficients. The mean and variance of the input speech signals are also included in the sideband information. The data of the sideband and mainband are encoded by the three block of a (63,45) BCH code in order to reduce the effects of the channel errors. The channel errors occurring during the transmission through the noisy channel will be corrected by the decoder. The information of the mainband is fed to the synthesizer to reproduce the speech signal.

2.3.2 Basis Spectrum of the ATC System

The performance of the ATC system is heavily dependent on the generation of the basis spectrum from which the adaptive quantization and dequantization rule is derived. Two basic adaptation techniques have been proposed. The first technique, proposed by Zelinski and Noll [1,2] is described in Figure 2.3-3.

After the calculation of DCT coefficients, the basis spectrum is estimated by making DCT coefficients positive and averaging between peaks to compress the DCT envelope. The amplitudes of the every mth (m is typically 8 to 16) sample of the envelope are quantized and sent to the receiver to represent the spectral levels at specified frequencies. These amplitudes are then geometrically interpolated (i.e., linearly interpolated in log amplitude) to form the basis spectrum. This simple, "non-speech

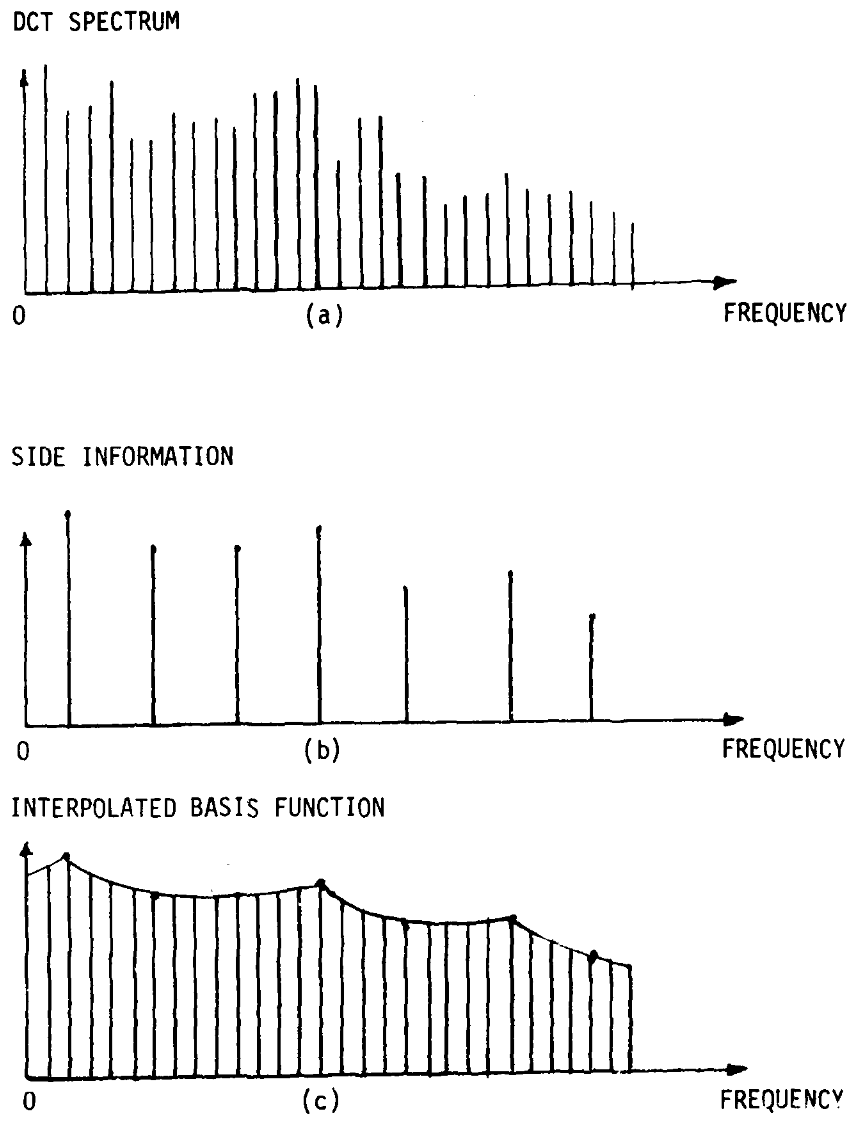


FIGURE 2.3-3 ESTIMATION OF THE BASIS FUNCTION

(a) Actual squared amplitudes of the DCT coefficients

(b) Averaged samples

(c) Estimated basis spectrum obtained by interpolation specific" algorithm is quite appropriate for speech transmission above 9.6 Kbps. However, the synthesized signal is degraded by a very perceptible "burbling" distortion as the data rate decreases below 9.6 Kbps.

Zelinski and Noll [2] suggested incorporating a form of voice-excited "fill-in" procedure similar to that used in voice excited vocoder technique. In their techinque, low energy frequency bands, which receive no bits for encoding at the transmitter, are filled-in at the receiver with random noise in order to enhance the perceived speech quality. Some improvements have been reported, but the addition of random noise introduces some hoarseness to the synthesized speech. They adjust the amount of added random noise to optimize the speech quality, i.e., the problems of "bubbling" and "hoarseness" are reduced, but it is not sufficient to overcome the difficulties aforementioned at data rates below 9.6 Kbps. Tribolet and Crochiere [3] proposed a more appropriate algorithm for bit rates below 9.6 Kb/s which is a "speech specific," adaptation algorithm, and takes full advantage of the known models and dynamics of the speech production mechanism in order to predict the DCT spectral levels. This algorithm is based on an all pole model of the formant structure of speech and a pitch model to represent the fine structure (pitch striations) in the speech spectrum [13] [14]. The resulting algorithm is referred to as a "vocoder-driven" adaptation strategy due to the close relationship of this spectral estimate to a vocoder model.

The block diagram in the Figure 2.3-1 illustrates the implementation of the technique. First the DCT spectrum is squared and inverse transformed with an inverse DFT. This yields an autocorrelation-like function, the pseudo-ACF (Auto-Correlation Function). The first P + 1 values of this function are used to define a correlation matrix in the usual normal equations formulation sense[13]. The solution of these equations yields an LPC filter of order P. The inverse spectrum, illustrated in Figure 2.2-3(c) yields an estimate of the formant structure of the DCT spectrum denoted as $\sigma_t'(k)$.

The fine structure of the DCT spectrum is obtained from a pitch model. To obtain the pitch period, M, the pseudo-ACF is searched for a maximum. The pitch estimate taken from the rudimentary procedure suggested by Tribolet and Crochiere[3,4] has a definite bearing on the SNR of the processed speech. The use of this imperfect pitch value does not grossly affect the subjective voice quality. However, in order to derive the most impact from the use of a pitch weighting function, the original pitch extraction procedure has been modified. The flowchart of the search routine for pitch period, M, is shown in Figure 2.3-4. It consists of a simple search routine which commences after the appearance of the second zero crossing in the autocorrelation function. The pitch contour which results from this technique is more accurate than the original unconstrained approach with a corresponding increase in the cumulative SNR. This simple scheme has proven to be adequate for the development of the ATC system when the voice/unvoiced decision device is incorporated. The corresponding pitch gain, G, is the ratio of the pseudo-ACF at M over its value at the origin. With these two parameters, a pitch pattern $\sigma_p(k)$ is generated in the frequency domain as illustrated in Figure 2.2-3(d). The two spectral components $\sigma_t(k)$ and $\sigma_p(k)$ are multiplied and normalized to yield the final spectral estimate for $\sigma_s(k)$,

$$\sigma_s(k) = \sigma_t(k)\sigma_p(k) \qquad k = 0, 1, 2, \ldots, N-1 \tag{2.3-1}$$

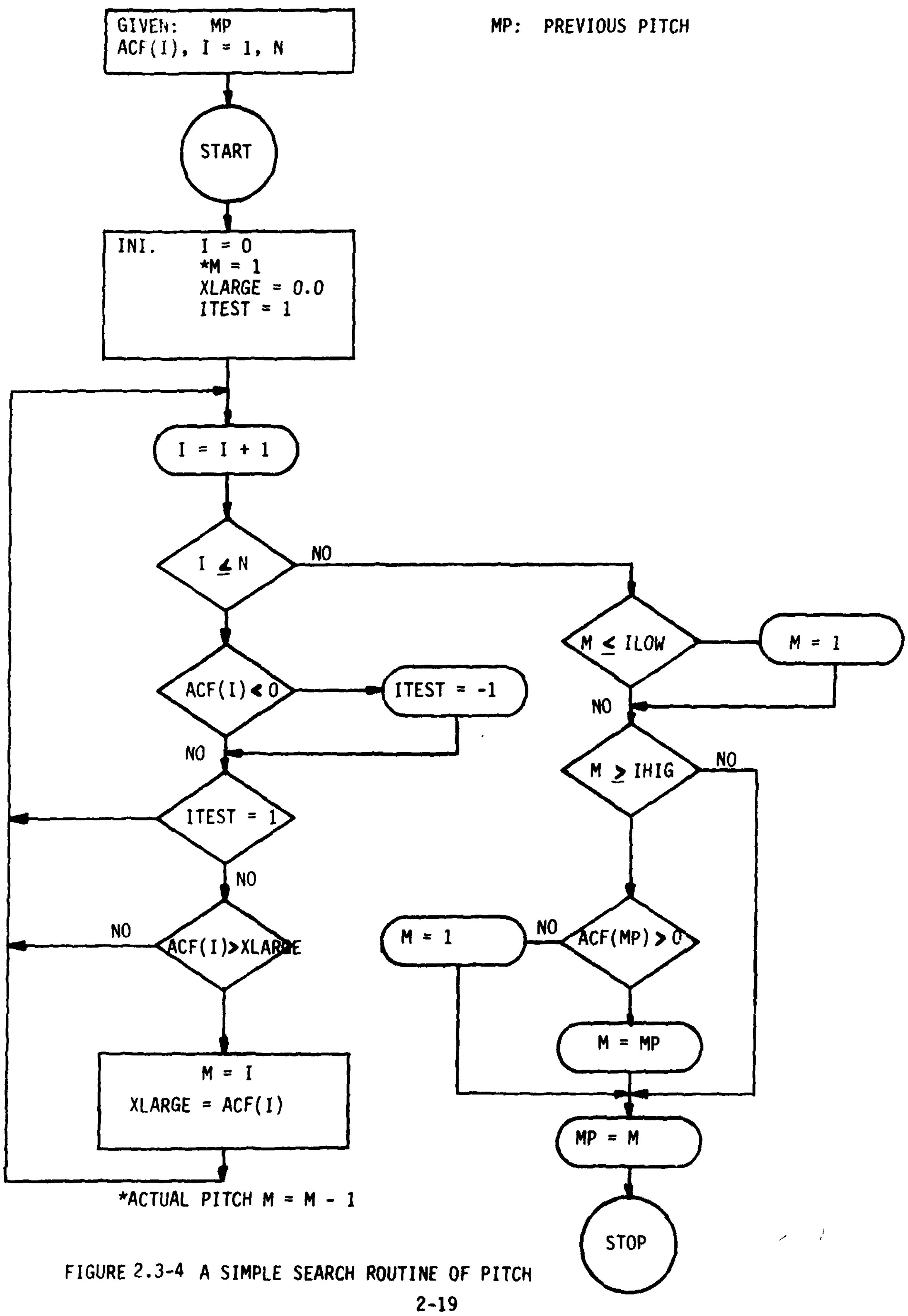


FIGURE 2.3-4 A SIMPLE SEARCH ROUTINE OF PITCH

This estimate, illustrated by Figure 2.2-3(e) is then used for the bit assignment and step-size adaptation algorithms as seen in Figure 2.3-1.

There are many ways of generating pitch weighting function in the frequency domain. In the model of GTE Sylvania, first the pitch gain is defined as

$$G = \mathrm{ACF}(M) / \mathrm{ACF}(0) \tag{2.3-2}$$

and a time domain pitch impulse train with exponentially decaying amplitudes is generated as

$$p(n) = \begin{cases} G^k & , \quad n = kM,\ k = 0, 1, \ldots, K, \quad K = \lfloor N/M \rfloor \\ 0 & \quad \text{otherwise} \end{cases} \tag{2.3-3}$$

where N is the number of speech samples in a frame and $\lfloor\cdot\rfloor$ denotes the largest integer. This time domain signal $p(n)$ is transformed into a zero mean and unit power process $p_1(n)$ which is again transformed into the frequency domain as

$$P_1(k) = \sum_{n=0}^{N-1} p_1(n)\, e^{-j\frac{2\pi kn}{N}} \quad , \quad k = 0, 1, \ldots, N-1 \tag{2.3-4}$$

This periodic pitch weighting function $P_1(k)$, shown in Figure 2.3-5, when multiplied by the LPC spectrum, is adequate for the generation of the basis function in many cases. However, there are cases in which the pitch harmonics are not well preserved in the high frequency band for some voiced sounds, particularly for the fricative voiced sounds (V, Z). There are also many cases where the pitch harmonics of $P_1(k)$ are not

MODIFIED PITCH WEIGHTING FUNCTION FOR VDS-ATC

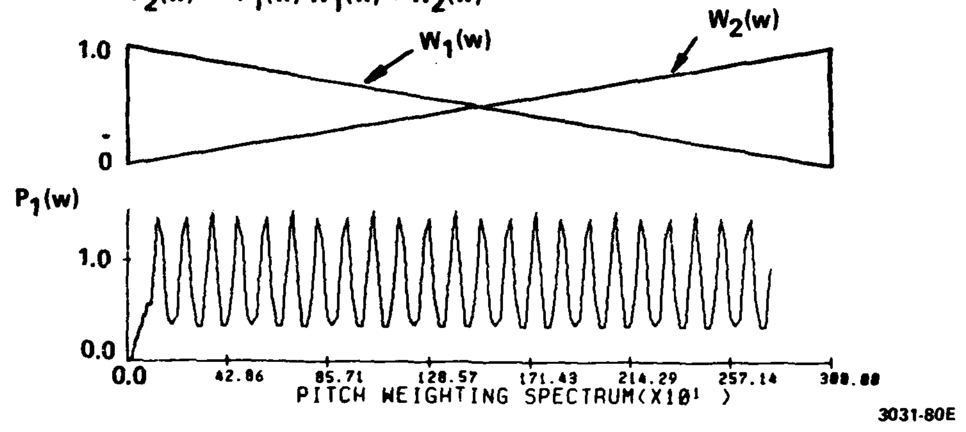
MODIFY THE PRIMARY IMPULSE RESPONSE, p(n), TO YIELD A ZERO VALUED DC COMPONENT AND UNITY POWER

$$p_1(kn) = p(kn) - \sum_{k=0}^{K} \frac{PG^k}{N}, \; K = \left[\frac{N}{M}\right]$$

$$P_1(w) = DFT_{2N}\left\{p_1(kn)\right\}$$

THEN APPLY COMPLIMENTARY LINEAR WEIGHTING FUNCTIONS, $\underline{W}_1(w)$ AND $\underline{W}_2(w)$ SUCH THAT

$$P_2(w) = P_1(w)\, W_1(w) + W_2(w)$$



FIGURF 2.3-5 GENERATION OF THE PITCH WEIGHTING FUNCTION

matched to the actual spectrum of the input speech, particularly in the high frequency band. This fact can be explained from the errors of the pitch period in time domain. Most gross errors of the pitch period are caused by erroneous decisions of the pitch detection routine. However, a small amount of erroneous pitch period estimates may always exist because of the discrete sampling process in the time domain.

Therefore, we generated a pitch weighting function which is periodic in the low frequency band and close to unit amplitude in high frequency band. One such function can be generated as

$$P_2(k) = P_1(k)\, W_1(k) + W_2(k) \tag{2.3-5}$$

where the weighting functions $W_1(k)$ and $W_2(k)$ are shown in Figure 2.3-5. The pitch weighting function $P_2(k)$, of equation (2.3-5), when it is multiplied by the LPC spectrum, has proven to be an efficient for the estimation of the basis function.

The basis function of the ATC system will be the LPC spectrum in eq. (2.3-1) with or without multiplication of the pitch weighting function. Experiments have shown that a closer estimation (in the mean square error sense) of the basis function to the actual spectrum makes the ATC system perform better. In fact, in the extreme case, where the basis spectrum equals the DCT spectrum, quantization of the DCT parameter can be precise, eliminating all types of distortion as long as the sign bits of the DCT coefficients are provided. A post V/UV decision is made on the basis of the signal-to-noise ratios in frequency domain with or without multiplication of the pitch weighting function, i.e., if the signal-to-noise ratio of the basis function without multiplication of the pitch weighting function provides higher value than the one with pitch weighting function, it is better to make that frame as unvoiced.

2.3.3 Bit Assignments Rule of the ATC System

It has been shown that the basis function of the ATC system plays an important role on the performance of the ATC system. The choice of bit assignments also determines how accurately the DCT coefficients are encoded. Thus, it controls the distribution of the quantizing noise in the frequency domain. The optimum bit assignements rule (in the minimum mean square error criterion) for a stationary Gaussian-Markov process has been derived in eq. (2.2-6) from the rate distortion theory.[15,16] It can be shown that the optimum bit assignments rule based on a minimum mean square error leads to a flat noise distribution in the frequency domain. It has been known that a flat noise in frequency domain is not the most desirable perceptual criterion. Tribolet and Crochiere[4] modified the bit assignment rule of eq. (2.2-6) by multiplying a weighting function W(k) that weights the importance of the noise in different frequency bands. They have suggested the weighting function to be

$$W(k) = \sigma_s^{2\gamma}(k), \quad k = 0, 1, \ldots, N-1 \tag{2.3-6}$$

where γ is a parameter that can be experimentally varied from -1 to 0. So the value of γ is slowly varied between these two extremes ($-1 < \gamma < 0$), the noise spectrum will evolve from a flat distribution to the one that precisely follows the speech spectrum. Extensive experiments[17,18,19] of noise shaping have shown that the noise spectrum which follows the spectrum of the speech in certain ways provides slightly higher subjective speech quality than the one of flat noise spectrum does. The value of $\gamma = -0.125$ was reported to give a good result[4]. However, when the data rate decreases below 8 Kb/s, the effects of the weighting function as well as the optimum bit assignments rule cannot be described clearly. The performance of the ATC system may be optimized asymtotically at the low data rates by incorporating a simple limiter of the highest bit allocation. By adjusting the largest number of the bit allocation, the spectrums of the noise can be varied. The spectrum of noise will be flat for the large value of the limiter output, and the spectrum of the noise will follow the speech spectrum when the value of the limiter output is small ($\gtrsim 1$). Experiments have shown that the maximum number of bit assignments of 5 provides a good result at the data rate 9.6 Kb/s.

2.3.4 Quantization of Sideband and Mainband Information

The quantization effects at high transmission data rates do not cause a perceptual loss of performance of the ATC system. However, at low data rates, these quantization effects constitute a major source of degradation of the synthesized speech.

Let P_j^2 be the jth actual spectrum and P_{sj}^2 be the jth spectrum from the side information. Then, the normalized DCT coefficient can be expressed as

$$x_j = P_j/P_{sj} \qquad (2.3\text{-}7)$$

Let B_j be the number of bits assigned to jth DCT coefficient. Then P_j is a Gaussian distributed random variable if the samples of time domain signals are Gaussian distributed. The distribution of the normalized DCT coefficients is, however, not known and analytical derivation of this distribution function is too complicated to calculate. GTE Sylvania performed simulations to develop the distribution function and the results are shown in Figure 2.3-6. Note that the distribution function of x_j lies in between the Gaussian and Laplace distribution. GTE Sylvania has written a computer program which calculates the characteristics of an optimum quantizer from the simulated distribution function by following the procedures of Max [20].

Let $\hat{x}_j$ be the quantized value of x_j, then the procedure of Max minimizes the mean square error, i.e.,

$$e = E|(\hat{x}_j - x_j)^2| \tag{2.3-8}$$

where E denotes the statistical expectation. GTE Sylvania has determined the distribution of x_j under the given conditions of B_j which is a function of the ATC coder and speech signals. The conditional distribution of x_j is slightly different from the distribution of Figure 2.3-6. GTE Sylvania has developed a computer program which generates an optimum quantizer from the conditional distribution of x_j. These procedures can be applied to develop quantizing tables for every system parameter where the minimum mean square error criterion is an adequate measure.

In the present ATC scheme, the LPC technique is used to calculate the basis spectrum with transmission requiring quantization of the PARCOR coefficients. In this case, the error criterion of eq. (2.3-8) is modified as

$$E = E|s(x_j)\ (\hat{x}_j - x_j)^2| \tag{2.3-9}$$

where s( ) is the weight function derived from the sensitivity analysis of power spectrum with respect to PARCOR coefficient variation [21]. A large data base was used to accumulate the statistical information needed for optimal quantization of the PARCOR parameters.

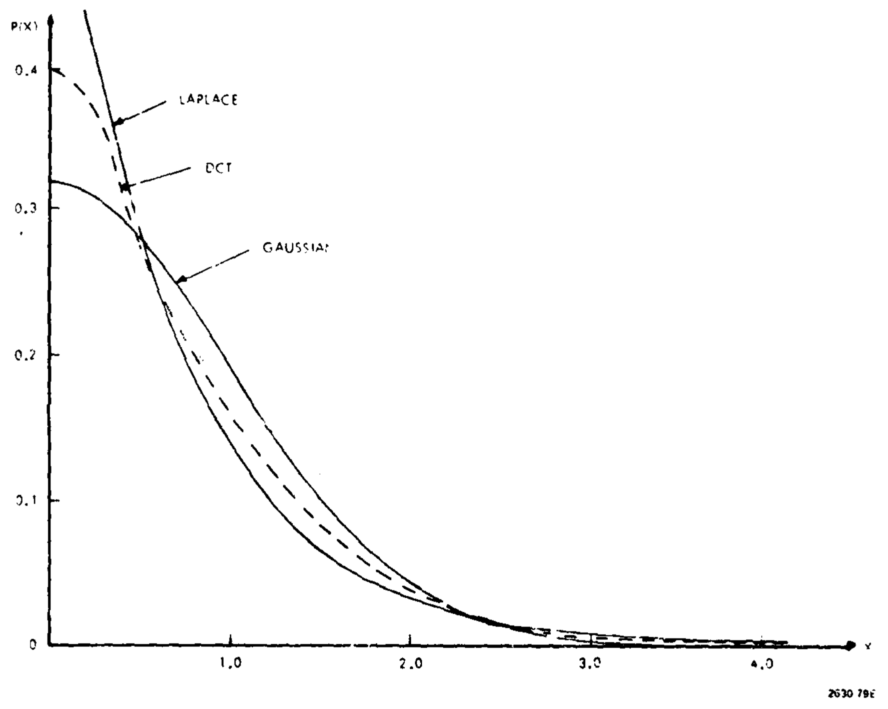


Figure 2.3-6 Probability Density Function of Discrete Cosine Transform Coefficient

The quantizer for each variable when optimized with the proper statistical error criterion, appears adequate for the ATC system over a variety of different speakers and acoustics noise conditions and data rates.

2.3.5 Reducing the Effects of Lowpass Filtering

The quality of the ATC coder degrades as the transmission data rate decreases. One of the major sources of this degradation is the low pass filtering effect which can be explained by examining Figure 2.3-7. This figure shows that no DCT coefficient is transmitted in frequency band 2. However, Figure 2.3-7 illustrates that the basis spectrum which is derived from LPC techniques closely follows the actual spectrum. The DCT coefficients of frequency band 1 are quantized from the LPC basis spectrum, where the phase (sign in this case) and amplitude information are modified from the LPC basis spectrum. This change results in an improvement over the LPC technique. In frequency band 2, the LPC basis spectrum cannot be modified since no bits are assigned. Zelinski and Noll [2], who use a different basis spectrum, substitute the DCT coefficients in this frequency band 2 with the noise samples whose variances are derived from the side information. Some improvements have been reported at low data rates.

This technique perceptually adds some bandwidth to the ATC system, but introduces some hoarseness to the speech. This hoarseness arises from the destruction of pitch harmonics in the frequency domain, and can be reduced by using the LPC basis spectrum modified by the pitch weighting function of eq. (2.3-5). This basis function is shown in Figure 2.3-7 with "#" symbol. The optimized ATC coder with the "fill in" procedure produces high quality synthesized speech above the data rate 7200 b/s.

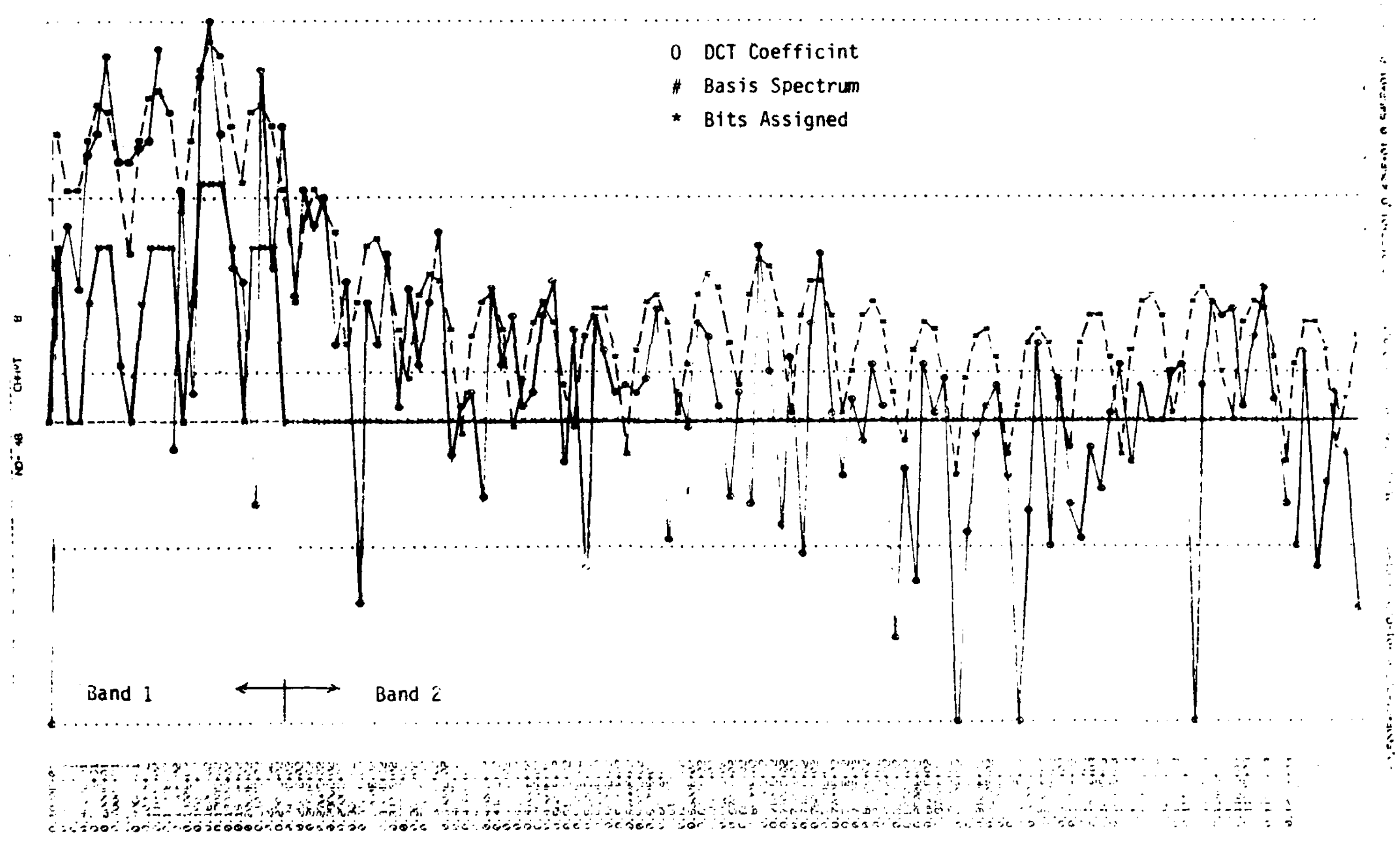


Figure 2.3-7: DCT, Basis Spectrum and Bits Assigned in ATC Coder

2.3.6 Bit Allocations to Sideband and Mainband

The performance of the ATC system depends on the several system devices, including quantizer characteristics, bit assignment rules, methods of estimating the basis function, bit allocations to the sideband and mainbands, etc. It has been shown that the estimation of the basis function plays an important role on the performance of the ATC system. However, it is desirable to allocate fewer bits for the generation of the basis function so that more bits remain for encoding DCT coefficients. Thus, tradeoff analyses were conducted in the area of DCT coefficient and LPC parameter quantization.

First, the performance of the ATC system was measured with the basis spectrums estimated by 10th order and 8th order LPC process (no quantization is applied to the PARCOR parameters). Both SNR measurements and informal listening tests have shown no significant differences. This is an important finding, since the quantization of this sideband information consumes a fair amount of the available data rate. Any conservation of bits in the LPC process can be used to improve or protect the transmission of the DCT coefficients.

Second, a large data base, comprised of 15 male and female speakers totalling 30,000 frames, was used to create relative frequency histograms for each PARCOR coefficient. The probability density functions were derived from this data and used with the technique described in section 2.3.4 to develop optimal quantizers.

The bit allocations strategies for the sideband information are shown in Table 2.3-1. Combining all the sideband information parameters, the total data rate is in the range of $35 \leq$ bits/frame ≤ 45. Since it is

Parameter	# of bits/frame	
PARCOR #	# of Bits Tested	# of bits Decided
1	4, 5	5
2	3, 4, 5	5
3	3, 4	4
4	3, 4	4
5	2, 3	3
6	2, 3	3
7	2, 3	2
8	2, 3	2
pitch, M	6	6
pitch gain, G	2, 3	2
variance	5	5
sync	1	1
V/UV	1	1

TABLE 2.3-1 BIT ALLOCATIONS TO THE SIDEBAND INFORMATION

necessary to allocate bits for the error correcting code in order to reduce the effects of channel errors, the data rate may be increased to $85 \leq$ bits/frame ≤ 95. By allocating 7200 bits/second for the encoding of DCT coefficients, there are 2400 b/s left for the sideband information. This limits the frame updating rate to less than 30 frames/second which forces the frame size of 256 samples with a 6400 Hz sampling rate. In order to reduce the effects of the signal discontinuities at the frame boundaries, the frames are overlapped slightly (10 samples). Therefore, there are 369 bits per frame (246 samples) with 6400 sampling rate for the 9.6 Kb/s ATC system.

With the above constraints, the performance of the 9.6 Kbps ATC system was evaluated with various combinations of bit allocations to the sideband information parameters. As a result, the combination of the bits sequence shown in Table 2.3-1 was determined to be optimal with respect to objective measurements (SNR). The performance of the ATC system is plotted with respect to the data rate in Figure 2.3-8 with the sideband data rate shown in Table 2.3-1. The figure shows that the decision on the sideband data rate is adequate for the ATC system of 6800 b/s ~ 9600 b/s, since its performance is not sensitive to the changes of the data rate.

2.3.7 Reducing Discontinuities at the Frame Boundary

The adaptive transform coding scheme of Figure 2.2-2 produces noise-like "burbling" and "click" sounds at low data rates. This noise is generated at the frame boundaries by waveshape discontinuities in the time domain. The noise generated from these discontinuities cannot be entirely eliminated, but can be reduced by overlapping frames slightly and by interpolating across the frame boundaries.

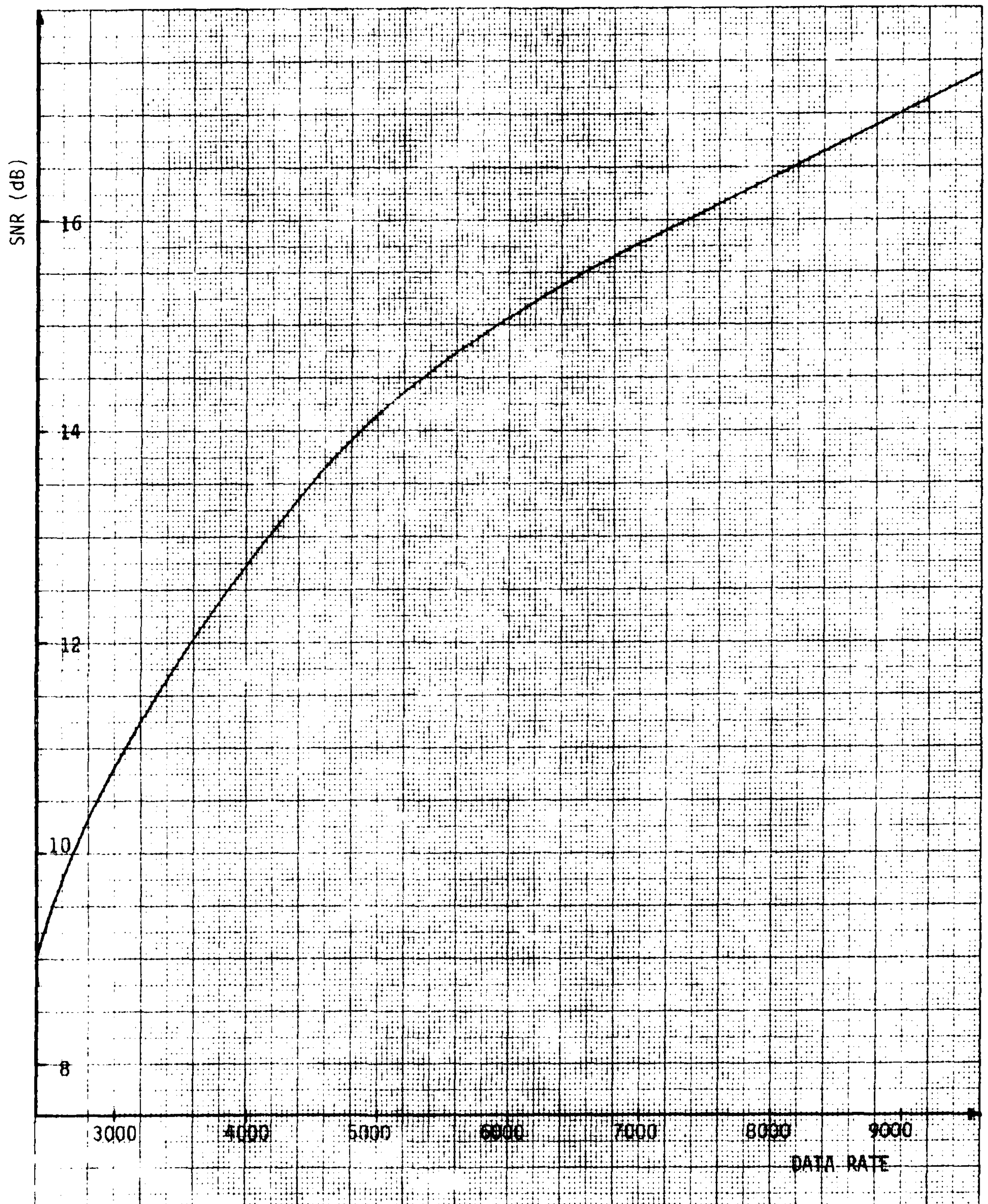


FIGURE 2.3-8 THE PERFORMANCE OF AN ATC SYSTEM

The frame size of the ATC system may be chosen as 128 samples/frame in the previous section. However, the frame size was increased to 256 samples to reduce the effects of signal discontinuities at the frame boundaries. The FORTRAN simulations of the ATC system at frame sizes of 128 and 256 samples revealed two beneficial findings. First, a larger frame size does not adversely affect the signal-to-noise ratio (SNR) but noticeably improves the subjective voice quality. Second, a larger frame size with pitch weighting is better than a smaller frame size with pitch weighting. Both these findings can be explained rather simply. The short term speech spectrum may not be stationary for a large frame size (246 samples), which may cause the synthesized spectrum to be smoothed more than it should be. However, since the frames are updated half as often, there are half as many frame discontinuities. In the ATC system, the frame discontinuities are the most obvious distortion and are lessened significantly with the 256 sample frame size. As the frame size increases, the resolution of the FFT increases as well due to an increase in the FFT order (N), i.e.,

$$\text{frequency resolution} = \frac{BW}{N} \quad , \quad BW = \text{signal bandwidth} \qquad (2.3\text{-}10)$$

A finer frequency resoltuion of the pitch weighting spectrum places more striations in the LPC basis spectrum. Hence, a more detailed sampling of the spectrum is achieved for larger frame sizes.

2.4 ATC System in the Presence of Random Channel Errors

Our FORTRAN simulations have shown that the ATC system produces high quality synthesized speech at 9600 bps if no channel errors are present. Error-free transmission, however, is not always possible since the transmitted signals are often affected by channel characteristics and by noise which may or may not vary in time. Additive Gaussian noise is the main source of signal corruption in many digital data transmission systems that will introduce random channel errors (error positions are independent of time). These channel errors may degrade speech quality, and the degradation of speech may depend on the positions of these channel errors.

In the following sections, the design of the ATC system will be examined and changed to optimize the performance of the system under the influence of random channel errors ranging from a bit error rate (BER) of 0 to 10^{-2}. First, the effects of random channel errors on the performance of the ATC system will be examined in section 2.4.1. Afterwards, the performance of the ATC system, as a function of data rate and channel error rate, will be provided in section 2.4.2. Then forward error correcting codes will be employed to reduce the effects of channel errors. The application of BCH codes, which are presently the most powerful random-error-correcting codes, will be presented in section 2.4.3. The performance of the ATC system is sensitive to the errors in the sideband information, since the bit assignments of the DCT coefficients depend on the sideband information. Some DCT coefficients are more important than the others in the sense of maintaining the system's performance with no channel errors. The selection and protection of the important bits in ATC system were made by analyzing the performance of the ATC system with various bits protected. Then, in section 2.4.4, we selected parameters of BCH code used to protect these bits. Finally, the conclusions are summarized in section 2.4.5.

2.4.1 The Effects of Random Channel Errors on the Performance of the ATC System at 9.6 Kbps

The ATC algorithm was originally designed for the error-free channel. A noisy channel, however, will introduce errors in the received bits. The performance of the ATC coder given by its signal-to-quantization noise ratio (SNR) is plotted in Figure 2.4-1 with respect to the channel error rate. These plots are obtained by evaluating various types of speech totally about 30 sec.

Degradation of the synthesized speech due to the channel errors was not noticed in the informal listening tests when the channel error rate was lower than 10^{-3}. However, the quality of the speech as well as the SNR drops rapidly when the channel error rate is higher than 10^{-3}. It is, therefore, desirable to protect the system performance at the higher channel error rates ($>10^{-3}$).

2.4.2 Tradeoff Analysis Between Data Rate and Channel Error Rate

The performance of the ATC system was evaluated in terms of the SNR with the transmission data rate varying from 7700 to 9600 bps. The results are plotted in Figure 2.4-2 together with the performance of the ATC system under the influence of random channel errors. As it is noted from this Figure, the performance of the system does not drop rapidly as the transmission data rate decreases, while the performance of the system drops rapidly when the channel error rate is higher than 10^{-3}. Since

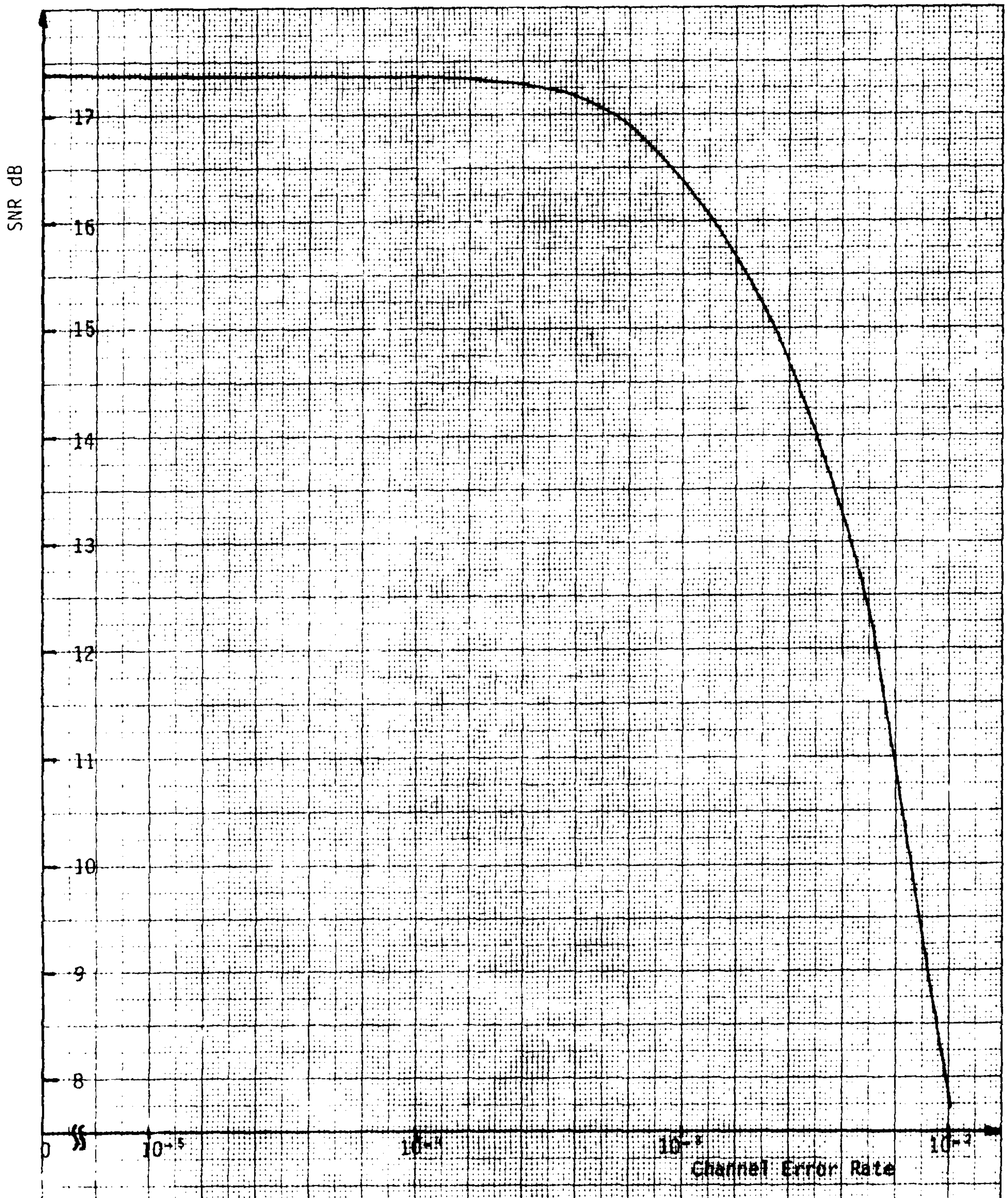


FIGURE 2.4-1: THE PERFORMANCE OF ATC SYSTEM UNDER NOISY CHANNEL

some channel errors can be corrected by utilizing error correcting codes, the error rate of the information bits can be reduced depending on the channel characteristics and the selection of error correcting codes. However, additional information (parity bits) has to be sent to the receiver to correct channel errors. Therefore, reducing the channel errors requires reducing the source information rate. Figure 2.4-2 shows that the performance of the ATC system can be improved by using the error correcting codes when the channel error rate is higher than 10^{-3} since the performance of the system drops slowly when the source information rate decreases. However, error correcting codes are not advisable to reduce the effects of channel errors when the error rate is lower than 10^{-3} since the performance of the system does not drop rapidly as the channel error rate increases.

2.4.3 Application of BCH Code

There are many ways of utilizing error correction codes to reduce the effects of channel errors. The method of correcting errors depends on the application, i.e., data rate, channel error rate, complexity, cost, etc. Since the ATC coder is designed for real-time implementation, error correcting code must not require a large time delay for correcting channel errors. Block codes of short length are suited to the real-time implementation of ATC algorithm. These codes require no additional time delay to process the error correcting algorithm if the length of the block code is less than the number of bits received in a frame period. There are many types of block codes that can be properly used depending on the channel characteristics.

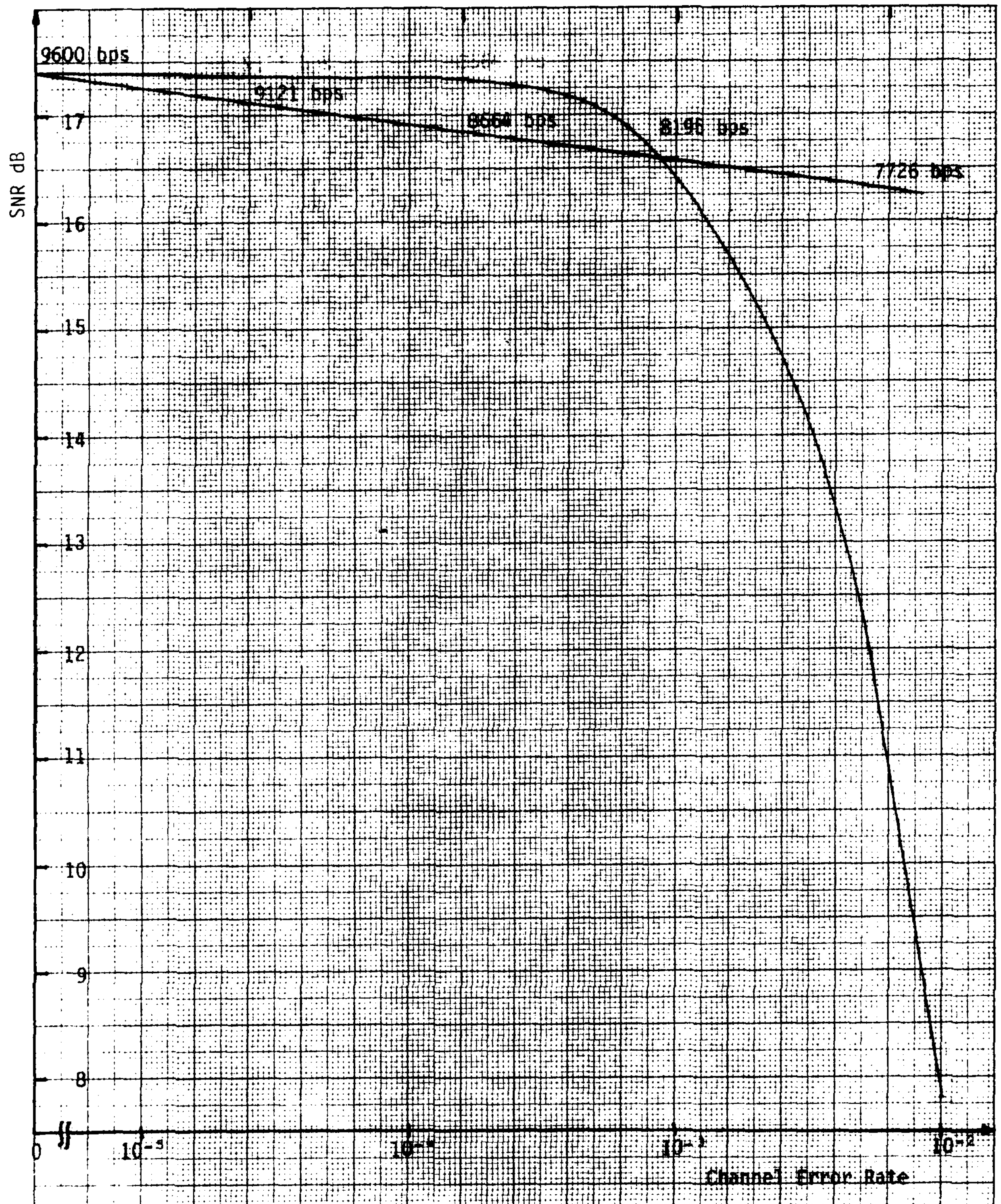


FIGURE 2.1-2: ATC PERFORMANCE VS. DATA RATE

Most real communication channels corrupt signals in many ways. The signals may be corrupted by the additive Gaussian noise and/or impulsive noise that produce random and burst errors, respectively. In other situations, the characteristics of the channel may vary in time (fading channel) or the channel may be selected at random from one ensemble of channels with widely different characteristics such as the switched telephone network. It is very hard to construct an error-control system which adapts to various types of channels. Since additive Gaussian noise is the main source that corrupt signals in many practical communication channels, the most practical error correcting code is the one which is capable of correcting random errors.

The Base-Chaudhuri-Hocquenhem (BCH) codes, that are a generalization of Hamming codes for correcting multiple errors. They are well known to be the most powerful random-error correcting codes, and a decoding algorithm that can be implemented with a reasonable amount of complexity has been devised for these codes.[10, 11, 12] A more fundamental description of the BCH codes and their encoding, decoding algorithm are given in Appendix A. This appendix shows that with the block length of the code $n=2^{m-1}$ and mt parity checks, it is possible to correct any t or less errors in a primitive (n,k) BCH code where k is the number of information bits. The proper choice of m, n, t in primitive BCH code depends on the channel error rate, data rate, and the system's specifications. For the real-time implementation of ATC coder, the values of t=3 and m=6, 7, 8 are considered as reasonable choices.

The performance of random-error correcting BCH codes is expressed in terms of error-probability. Let P(m,n) be the probability of m errors occurring in an n-bit block and β_m denote the probability of decoding an error pattern of weight m correctly, then the probability of decoding the received code word erroneously may be expressed as

$$P = 1 - \sum_{m=0}^{n} \beta_m P(m,n)$$

$$= \sum_{m=0}^{n} \alpha_m P(m,n) \qquad (2.4\text{-}1)$$

where $\alpha_m = 1-\beta_m$ denotes the probability of erroneously decoding an error pattern of weight m. The parameter α_m is a function of the code and decoding algorithm. If a t error-correcting BCH code is employed and it is decoded with the Peterson decoding algorithm shown in Appendix A, the parameter α_m may be expressed as

$$\alpha_m \begin{cases} = 0 & 0 \le m \le t \\ = 1 & t < m \le n \end{cases} \qquad (2.4\text{-}2)$$

and the probability of erroneously decoding the code word may be reduced from eq. (2.4-1) as

$$P_e = \sum_{m=t+1}^{n} p(m,n) \qquad (2.4\text{-}3)$$

If the bit errors occur independently and at random with probability e, then the probability p(m,n) can be expressed as

$$p(m,n) = \binom{n}{m} e^m(1-e)^{n-m} \qquad (2.4\text{-}4)$$

where the probability p(m,n) is simply the binomial distribution and P_e in eq. (2.4-3) is simply the tail of the distribution.

Let the channel error rate $e = 10^{-2}$ which is specified by the contract. Let the block length of the BCH code $n = 127$ and $t = 3$, then the probability of error occurring in the block can be written as

$$Pe = \sum_{m=4}^{127} p(m,127)$$

$$= 1 - p(0,127) - p(1,127) - p(2,127) - p(3,127)$$

$$= 0.0393 \qquad (2.4\text{-}5)$$

In this BCH code, the information rate may be expressed as

$$R = k/n$$

$$= 106/127$$

$$= 0.8346 \qquad (2.4\text{-}6)$$

where 16.54% of the data is used for the redundant parity checks. The information rate for the (127,106) BCH code from eq. (2.4-6) is a high 83.46%. However, the probability of error occurring in the block is also high since from eq. (2.4-5), it is expected to have one block in error for each 25 blocks. Let $n = 63$, $k = 45$, $t = 3$ ((63,45) BCH Code), then the probability of error occurring in the block of 63 bits will be

$$Pe = \sum_{m=4}^{63} p(m,63)$$

$$= 1 - p(0,63) - p(1,63) - p(2,63) - p(3\text{-}63)$$

$$= 0.003725 \qquad (2.4\text{-}7)$$

The information rate R is about 71.42%, which is lower than one of the (127,106) BCH code, but one erroneous block is expected out of 268 blocks, which turns out to be a proper choice in the following section.

2.4.4 Selection and Protection of the Important Bits in the ATC System

The performance of the ATC system was evaluated under various simulated channel error rates (random errors) to see the effects of channel errors. Two independent error generators were used on separate regions of the bit allocation strategies to isolate the most sensitive bits out of the 9600 bps system. One error rate, error rate A, was applied to the bit stream of the DCT coefficients. The other error rate, error rate B, was applied to the bits allocated to the sideband information parameters. The results are plotted in Figure 2.4-3. In this figure, the cumulative SNR does not degrade more than 9 dB when the channel errors are applied to every bit at the rate 10^{-2}. Although the processed sentence is generally intelligible, there are periods when concentrated errors lead to undesirable distortions (pops, clicks, etc.). On the other hand, a few bits of protection on the sideband information (42 bits/frame in a 369 bits/frame) lead to the improvement of SNR by 4.2 dB. Therefore, protection of the sideband information from the channel errors is necessary to minimize the reduction in SNR, since the performance of the ATC system is very sensitive to the errors in the sideband information.

To further improve the system's performance at error rates less than 10^{-2}, a primitive BCH code was applied for the partial protection of the bits related to the DCT coefficients as well as for the protection of sideband information. The protection

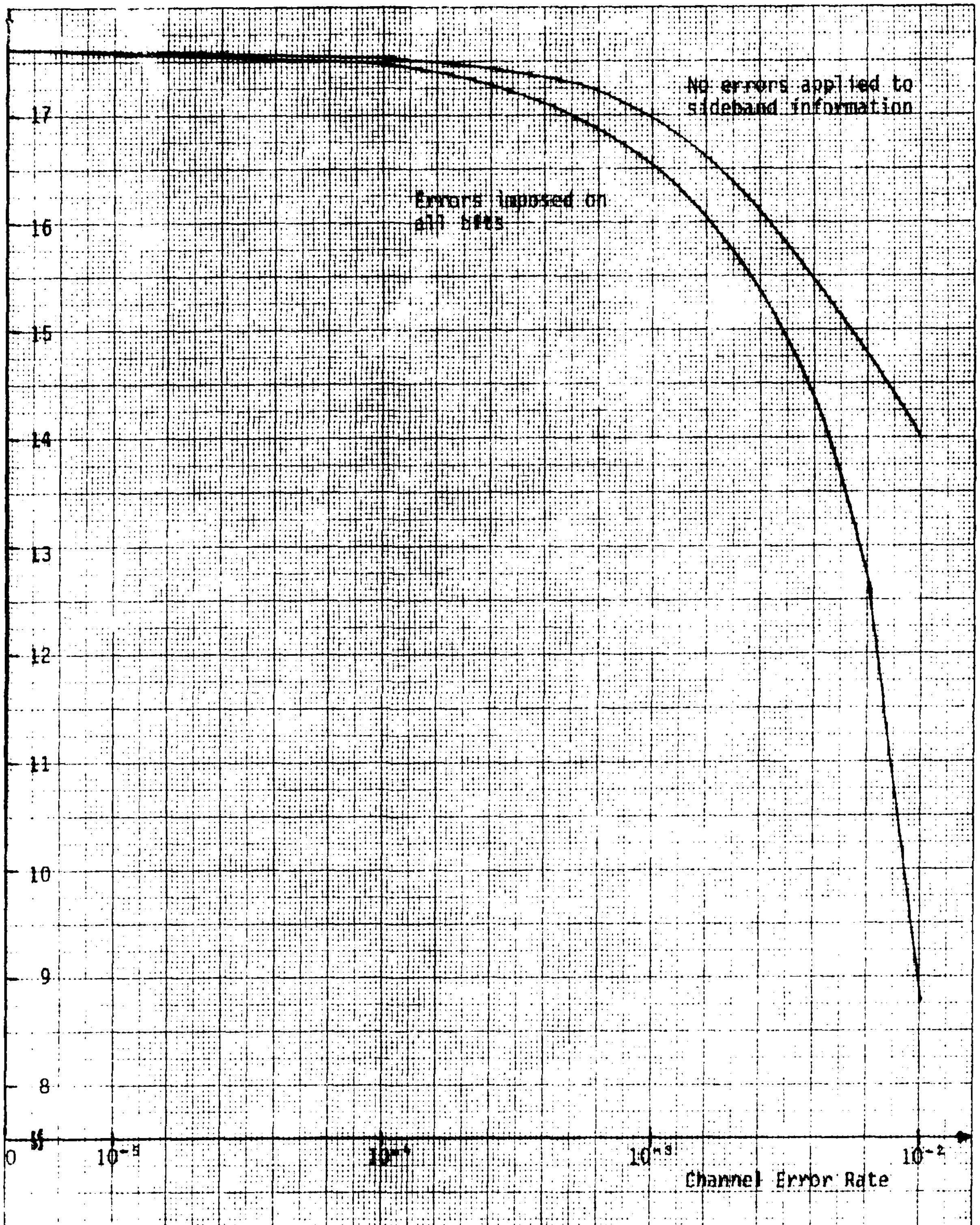


FIGURE 2.4-3: PERFORMANCE OF THE ATC SYSTEM WITH VARIOUS CHANNEL IMPAIRMENTS UPON DIGITAL DATA STREAM

of all information is not considered practical because of software requirements (computation time and program size), and as Figure 2.4-5 shows, the protection of all information does not lead to the highest system performance at the error rates below 10^{-2}.

In the early simulations of ATC system with channel errors by Zelinski and Noll,[1, 2] no channel errors are applied to the sideband information or to the most significant bit of each DCT coefficient. The performance of the system was shown to be insensitive to the changes of channel error rates up to 5%. Although the ATC system has been modified, the protection of the most significant bit from each DCT coefficient may lead to a good selection of important bits. We modified this scheme shown in Figure 2.4-4 (diagonal protections). In this figure, the quantized DCT coefficients (not the value of the quantized DCT, but the decimal number or address of the quantization tables which is ready to serialize for the transmission through the channel) are ordered in descending magnitude. The information on magnitude and the number of bits for each DCT is obtained from the sideband information. In the "diagonal protections," the bits to be protected are selected in the sequence of $1\rightarrow6\rightarrow10\rightarrow14\rightarrow17\rightarrow$ --- up to the desired number of bits. Another way of selecting important bits is the technique of "horizontal protections" where the bits are selected for protection in the horizontal direction in Figure 2.4-4 as $1\rightarrow2\rightarrow6\rightarrow10\rightarrow3\rightarrow7\rightarrow$ up to the desired number of bits. In order to select the number of bits to be protected, two block lengths of BCH codes (i.e., (63,45) and 127,106) BCH code) have been applied to the ATC system with the selections of important bits described as "horizontal protections" and "diagonal protections." The number of bits to be protected as well as the performance of the ATC system are tabulated in Table 2.4-1 at the channel error rate

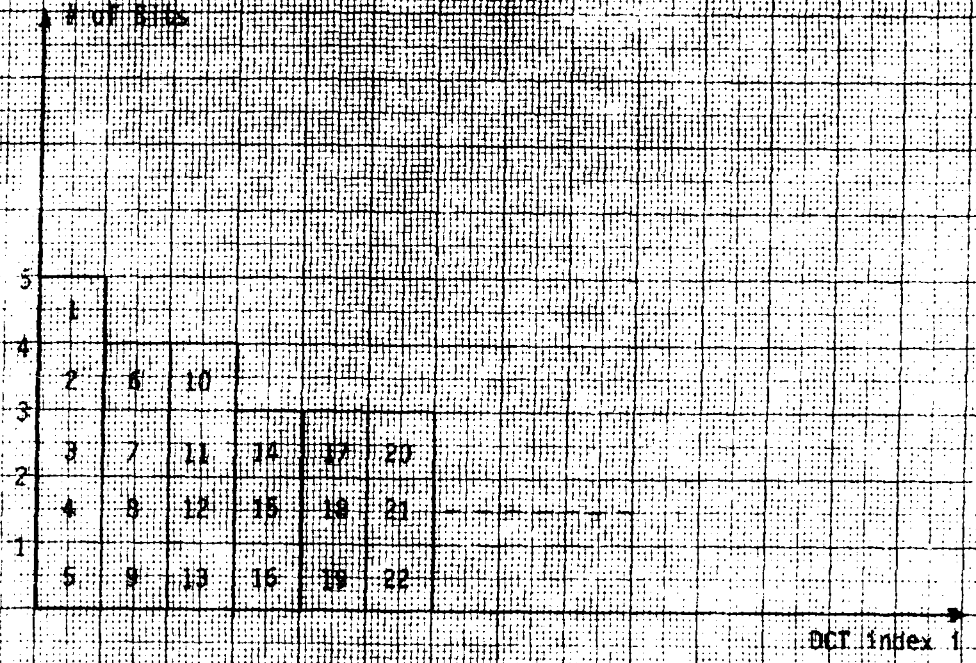


FIGURE 2.4-4: BITS ASSIGNMENTS OF DCT COEFFICIENTS IN DESCENDING ORDER

Program Name	Number of Bits Protected	Number of Bits Overhead	Name of BCH Code Used 2 times	SNR at Channel Error Rate 0	SNR at Channel Error Rate at 10^{-7}
ATC	0	0	*	17.3 dB	8.8 dB
ATCD Diagonal Protection	45 x 2 90	18 x 2 36	(63,45)	15.7	13.7
	106 x 2 212	21 x 2 42	(127,106)	15.6	14.4
ATCH Horizontal Protection	45 x 2 90	18 x 2 36	(63,45)	15.7	14.5
	106 x 2 212	21 x 2 42	(127,106)	15.6	15.7

*Total Number of Bits per Frame = 369 Bits

TABLE 2.4-1: PERFORMANCE OF ATC SYSTEM UNDER VARIOUS CHANNEL CONDITIONS

0 and 10^{-2}. As it is noted from the table, "horizontal protection" performs better than "diagonal protection" by 1.3 dB when two blocks of a (127,106) BCH code are applied to the ATC system at the channel error rate 10^{-2}. Another observation is that the (127,106) BCH code improves performance over a (63,45) BCH code by 1.1 dB in "horizontal protection" at the channel error rate 10^{-2}. However, informal listening tests indicate that there are periods that contain a large amount of distortions (pops, clicks, etc.) which lead to major objectionable speech degradation at 9600 bps. The main source of this degradation are the frames of speech that have more than 3 errors which cannot be corrected by the system. The probability of more than 3 errors occurring in a block is 0.039 from eq. (2.4-4) when the (127,106) BCH code is employed for the channel error corrections of rate 10^{-2}. This probability is reduced to 0.0037 when the (63,45) BCH code is incorporated. Thus, to reduce the number of frames that contain a large amount of distortions, one should use the (63,45) BCH code rather than the (127,106) BCH code when the channel error rate is 10^{-2}.

Finally, the Table 2.4-1 indicates that it may be better to protect more than 90 bits out of a frame (369 bits/frame) to increase the SNR at the error rate 10^{-2}. To protect more bits, one must use more parity bits which reduces the number of bits for encoding speech signals. The tradeoff analysis between the number of parity bits and error rates on the performance of the ATC system was investigated by using a (63,45) BCH code coupled with selecting the important bits by the technique of "horizontal protection."

To find out the best number of bits to be protected, the performances of the ATC system were evaluated at the several different channel error rates by varying the number of blocks for (63,45) BCH code in a frame. The results are tabulated in Table 2.4-2. The performance of the system is plotted in Figure 2.4-5 when the channel error rate is fixed at several values and the number of blocks of a (63,45) BCH code is a variable from 0 to 5. The performance of the system is also plotted in Figure 2.4-6 when the number of blocks protected from channel errors is fixed at some values and the channel error rate varies from 0 to 10^{-2}. As it is noted from Figure 2.4-5, the performance of the ATC system improves rapidly as the number of blocks (or number of bits) to be protected increases at the channel error rate 10^{-2}. As the number of blocks reaches 3, the performance of the ATC system saturates, while the best performance (denoted by Δ in the Figure 2.4-5) is obtained when 4 blocks of a (63,45) BCH code are employed at an error rate of 10^{-2}. At the channel error rate 5×10^{-3}, the best performance is obtained when the 3 blocks of a (63,45) BCH code are applied to protect the important bits of the ATC system. At the channel error rate 10^{-3}, the protection of one block (45 bits) is sufficient to obtain the best performance of the system. It is not necessary to protect any bits in order to reduce the effects of channel errors when the error rate is lower than 5×10^{-4}.

For the real-time implementation of the ATC system, the recommendation was to use the 3 blocks of a (63,45) BCH code because of the computation time and the saturation of the ATC system's performance. This conclusion was reached from the Figure 2.4-6, since the performance of the ATC system, when 3 blocks of a (63,45) BCH code is employed to reduce the effects of channel errors, is consistent and high for the channel error rates below 10^{-2}. The degradation of the performance due to the channel errors of rate up to 10^{-2} is less than 0.76 dB. Informal listening tests indicate

# of Blocks Protected \ Error Rate	0.0	10^{-4}	5×10^{-4}	10^{-3}	5×10^{-3}	10^{-2}
0 (0*)	17.39 dB	17.34	17.07	16.37	12.35	7.80
1 (45)	17.15	17.11	16.95	16.75	15.01	13.33
2 (90)	16.84	16.82	16.73	16.64	16.01	15.19
3 (135)	16.59	16.58	16.56	16.53	16.27	15.83
4 (180)	16.30	16.30	16.30	16.28	16.19	15.95
5 (225)	15.96	15.96	15.96	15.95	15.92	15.81

*The number inside the parentheses indicate the number of bits protected

TABLE 2.4-2 PERFORMANCE OF THE ATC SYSTEM WITH VARIOUS CHANNEL CONDITIONS

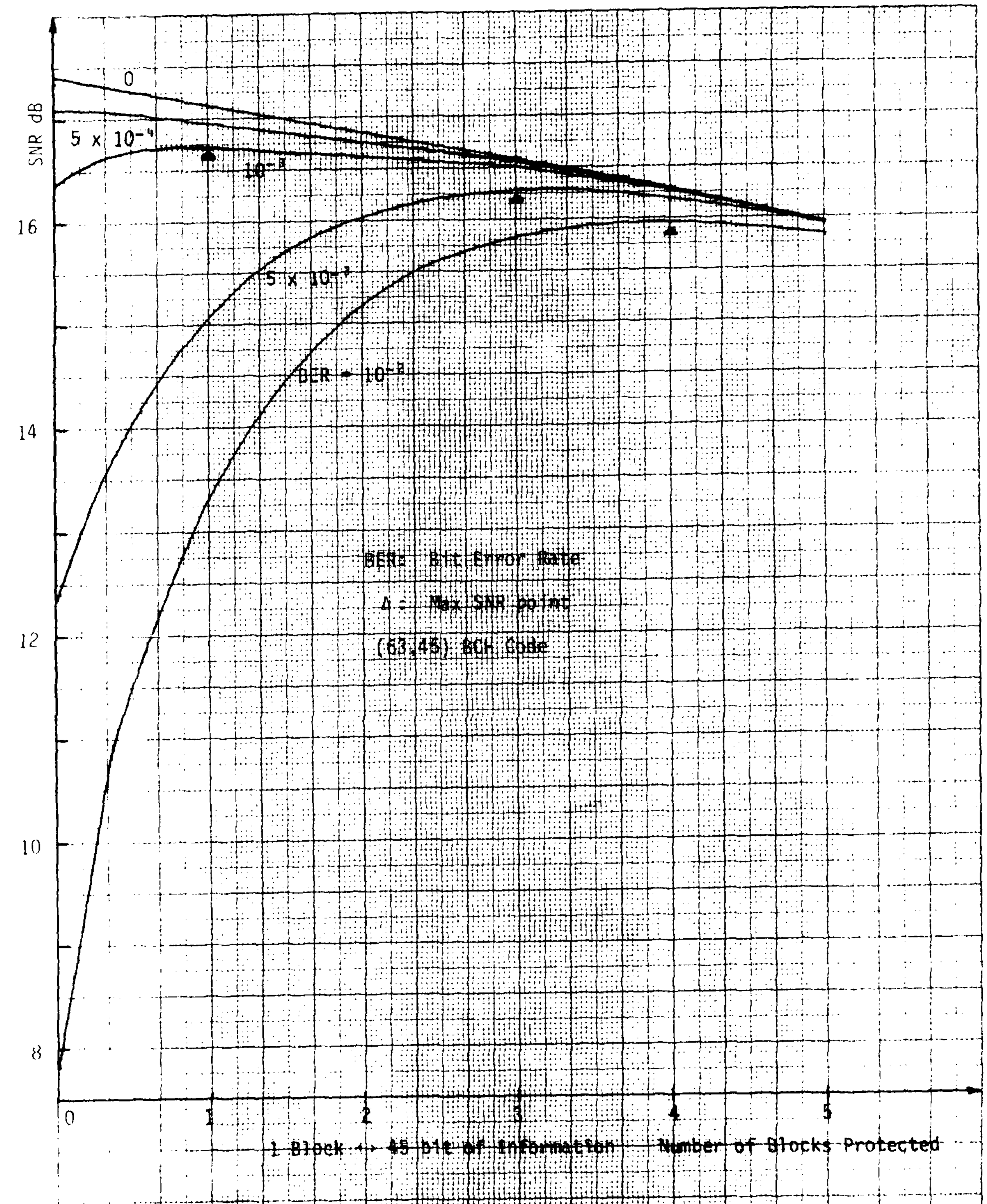


FIGURE 2.4-8: PERFORMANCE OF THE ATC SYSTEM VS. NUMBER OF BLOCKS PROTECTED

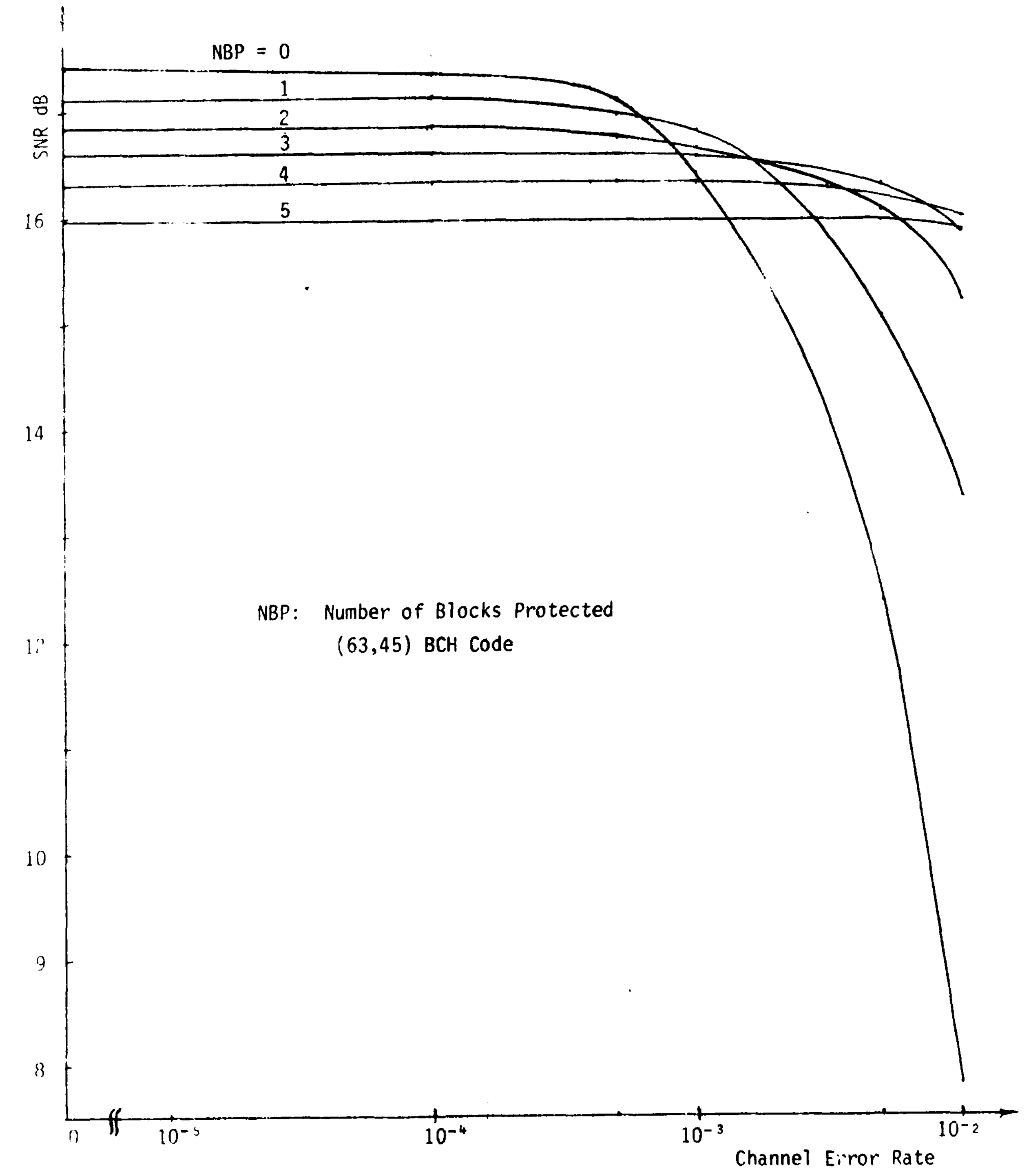


FIGURE 2.4-6: PERFORMANCE OF THE ATC SYSTEM VS. CHANNEL ERROR RATES

that the speech quality of 9600 bps is high and consistent in the presence of channel errors of rate up to 10^{-2}.

2.4.5 Summary

The ATC system had its transmission data sent through a Gaussian noise channel to investigate effects of random channel errors. The degradation of the speech quality or the signal-to-quantization noise ratio does not degrade significantly when the channel error rate is lower than 10^{-3}. However, the degradation of speech quality, when the channel error rate is higher than 10^{-3}, is so severe that one must reduce the effects of the channel errors. Since the performance of the ATC system has been insensitive with respect to the changes of the source information data rate, it was possible to employ the error correcting code to reduce the effects of the channel errors.

BCH codes were briefly described and were used to improve performance with channel errors. The selection and protection of the important bits in the ATC system were conducted by utilizing small block lengths (i.e., 63 or 127) of BCH code which correct up to 3 errors in the block. As a result, the selection of the important bits in ATC system were made by the technique of "horizontal protections." The (63,45) BCH code was also selected to reduce the number of periods which contain a large amount of signal distortion caused by a large number of burst errors (≥ 4) in the protected block.

The optimum performance of the ATC system was obtained for a given channel error rate. For example, the optimum performance of the ATC system (15.95 dB of SNR) was obtained when the 4 blocks of a (63,45) BCH code are incorporated to reduce the effects of the channel errors of the rate 10^{-2}. However, we recommended using 3 blocks of a (63,45) BCH code for the protection of channel errors up to the rate 10^{-2} because of the saturation of the ATC system's performance and real-time computation capability.

Finally, based on the SNR performance, we have shown that the ATC system designed in this study produces a high and consistent quality of speech at the data rate 9600 bps in the presence of random channel errors up to the rate 10^{-2}.

2.5 FORTRAN Program for the Simulation of the ATC System

The ATC scheme developed in the previous sections is programmed in FORTRAN, and the simulations of the ATC scheme are performed by a PDP-11 computer with a RSX-11M operating system.

The FORTRAN program will be described first in section 2.5.1. The task building of the program from the source file and the operation of the program will be shown in section 2.5.2. Appendix C contains a source listing of all FORTRAN programs for the ATC simulation.

2.5.1 FORTRAN Program of the ATC Algorithm

The ATC algorithm was developed using the FORTRAN programs before the real-time implementation of the ATC scheme began on the MAP-300 of CSPI, Inc. The flow diagram of the algorithm is shown in Figure 2.5-1. The programs consist of the main routine (ATC 70) and 25 subroutines. The functional descriptions of the program will be given following the flow diagram of Figure 2.5-1.

First, the parameters of the ATC system are defined in the initial setup routine within ATC 70. The quantizer tables for the sideband parameters and DCT coefficients are also defined in this routine. The use of the pitch weighting function, the sorting techniques (fast but approximate or slow but exact), input speech sampling rate, simulation channel error rate, and the information of the input/output speech file (name of the file and the storage device of the system, etc.) are defined in the subroutine OVR2. The characteristics of the ATC system are defined here and the program is ready to execute over and over until it is terminated.

The number of frames that are processed by the program is updated in the main program (labeled M1 on Figure 2.5-1) and the input speech is read and buffered by the subroutine TAPE3. The mean and variance of the input

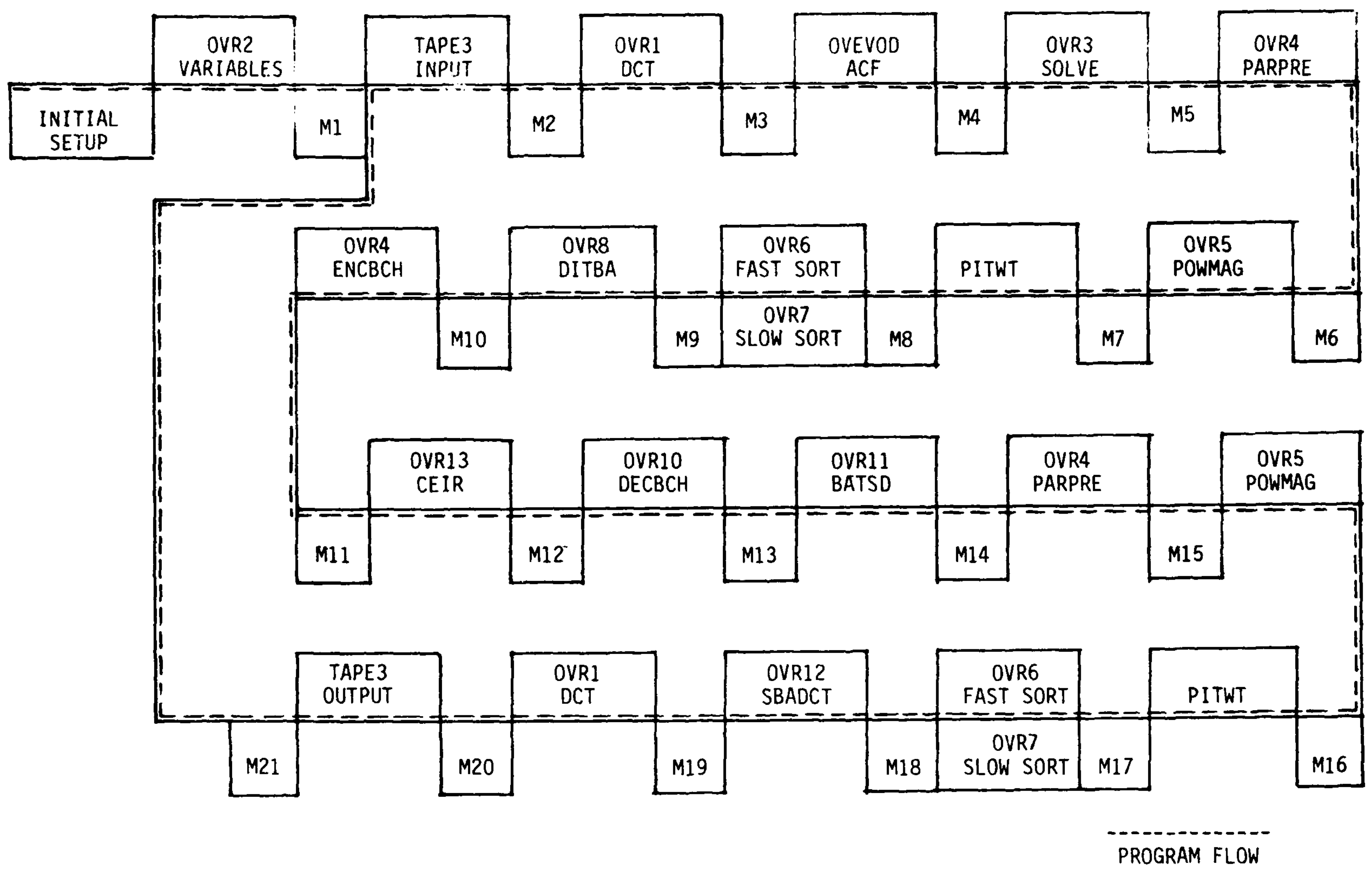


FIGURE 2.5-1: FLOW DIAGRAM OF THE ATC FORTRAN PROGRAM

signal are calculated for the normalization in the main program at M2, and the discrete cosine transform (DCT) is performed on the normalized input signal in the subroutine OVR1. The vectors are shuffled in the main routine at M3 for the fast calculation of the pseudo autocorrelation function (ACF) of the input speech signal which is performed in the subroutine OVEVOD.

The pseudo-ACF is searched for a maximum to obtain the pitch period, M. The corresponding pitch gain, G, is the ratio of the pseudo-ACF at M over its value at the origin. The pitch period, M, and the pitch gain, G, are determined in the main routine at M4. The PARCOR coefficients are calculated from the normalized pseudo-ACF in the subroutine OVR3. The quantizations and dequantizations of the PARCOR coefficients and pitch period, pitch gain, DC bias and variance of the input speech signal are performed in the main program at M5. The LPC filter coefficients are calculated from the PARCOR coefficients in the subroutine OVR4, and the generation of the LPC excitation source is performed in the main program at M6. DFT is performed on the LPC excitation source to get the LPC basis spectrum in the subroutine OVR5, and this spectrum is normalized in the main routine at M7. The pitch weighting function is generated in the subroutine PITWT, and it is multiplied to the LPC basis spectrum to form the ATC basis spectrum in the main program at M8.

The bit assignments rule is derived from the basis spectrum and the quantizations of the DCT coefficients are performed. First, the basis spectrum is sorted in descending magnitude in the subroutine OVR6 (fast sorting routine) or OVR7 depending on the terminal input. The fast sorting routine may provide an approximate result of the slow but exact sorting routine OVR7. The bit assignments routine and the quantizations of the DCT coefficients are performed in the main program at M9. The quantized decimal inputs are serialized into binary vector in the subroutine OVR8, and the encoding of a (63,45) BCH code is performed in the subroutine OVR9.

The encoded binary data is then transmitted through the simulated noisy channel in which the information of the transmitter may be altered due to the introduction of the channel errors. Simple tests are performed in the main routines at M10, M11, M12, and M13.

At the receiver side, the received sequence of binary data is fed to the decoder routine for the correction of errors if any in the subroutine OVR10. The sideband information is obtained first by unpacking the corrected binary vector in the subroutine OVR11. The sideband information, which consists of PARCOR coefficients, pitch period, pitch gain, mean and variance of the input signal, is dequantized in the main routine at M14. In order to generate the basis spectrum, LPC filter coefficients are calculated from the PARCOR coefficients in subroutine OVR4, and the time domain exitation source for the LPC spectrum is performed in the main routine at M15. The LPC spectrum is generated in the subroutine OVR5, and the pitch weighting function is calculated in the subroutine PITWT for the case of voiced sounds. The ATC basis spectrum is obtained by the multiplication of the LPC basis spectrum and pitch weighting function in the main program at M16 and M17. The basis spectrum is again sorted in descending magnitude in the subroutine OVR6 or OVR7. The bit assignments rule is exercised again from the sorted basis spectrum in the main program at M18. The mainband information is obtained by unpacking the received binary data in the subroutine OVR12. The dequantizations of the DCT coefficients are performed in the main program at M19, and the inverse DCT is performed to reproduce the time domain signal in the subroutine OVR1.

The time domain signal is renormalized by the mean and variance of the input signals and interpolated in order to reduce the effects of the signal discontinuities at the frame boundaries in the main routine at M20. This reproduced signal is fed to the output device in the subroutine TAPE3, and the post analysis (measuring the signal-to-noise ratio) is performed in the main program at M21. These procedures are repeated until the desired number of frames are processed by the program.

2.5.2 Task Building of the ATC Program

A magnetic tape was sent to DCA containing all of the source files necessary to build the ATC program.

Also included was 10 frames of data with zero and one percent error rates, that are shown in Figure 2.5-2 and Figure 2.5-3, respectively. The task module of the ATC program can be built as follows:

I) CATC70.CMD is an indirect command file that compiles all source needed for taskbuilding the overlay ATC program. It also purges all old object files. It also spools the overlay descriptor language program.

*Note that ATC70 is compiled with the slash DE option which allows for printout to LUN 4 all diagnostics in the ATC main program. This requires a larger compiler partition and also requires assignment of LUN 4 to a system device upon installation of the main program. Therefore, if diagnostics are undesired, do not compile with the /DE option. However, there is a rather elegant set of diagnostics, not to be passed over in haste.

This program is invoked by typing (in MCR)

```
@ CATC.70
```

11) ATCOLA.CMD task builds the overlay descriptor program creating the task ATC70. If a map is desired, then add the LP option in the BUILD.CMD program.

This program is invoked by typing (in MCR)

```
  TKB @ ATCOLA
```

The program executes by typing

```
  RUN ATC70
```

as shown in Figure 2.5-2.

```
[illegible]
SHALL WE DO SERIALIZATION OF DATA(Y/N)Y
SHALL WE INSERT ERRORS IN CHANNEL(Y/N)?Y
ERROR RATE IN E11.4=
[illegible]
SHALL WE ENCCODE AND DECODE(Y/N)?Y
USE PITCH WEIGHTING(Y/N)? Y
USE FAST SORT(Y/N)? Y
SAMPLING RATE AND XMIT DATA RATE IN 2I6=6400,9600
IS THE INPUT ON MAG. TAPE? N
IS THE OUTPUT GOING TO MAG TAPE? N
OUTPUT FILE NAME= NL:
INPUT FILE NAME= SPEECH.DAT
NO. FRAMES? 10
TOTAL NUMBER OF BITS=    369
INITIAL FRAME?5
FRAME=   5    SN= 22.37    CSN= 22.37    PITCH=  1    F.GAIN=   1.000    DCT BITS= 276

FRAME=   6    SN= 13.77    CSN= 18.07    PITCH=  1    F.GAIN=   1.000    DCT BITS= 276

FRAME=   7    SN= 22.12    CSN= 19.42    PITCH=  1    F.GAIN=   1.000    DCT BITS= 276

FRAME=   8    SN= 16.59    CSN= 18.71    PITCH=  1    F.GAIN=   1.000    DCT BITS= 276

FRAME=   9    SN= 17.08    CSN= 18.39    PITCH=  1    F.GAIN=   1.000    DCT BITS= 276

FRAME=  10    SN= 15.99    CSN= 17.99    PITCH=  1    F.GAIN=   1.000    DCT BITS= 276

FRAME=  11    SN= 16.55    CSN= 17.78    PITCH=  1    F.GAIN=   1.000    DCT BITS= 276

FRAME=  12    SN= 19.24    CSN= 17.97    PITCH=  1    F.GAIN=   1.000    DCT BITS= 276

FRAME=  13    SN= 23.89    CSN= 18.62    PITCH=  1    F.GAIN=   1.000    DCT BITS= 276

FRAME=  14    SN= 23.21    CSN= 19.08    PITCH=  1    F.GAIN=   1.000    DCT BITS= 276

FRAME=  15    SN= 16.59    CSN= 18.86    PITCH=  1    F.GAIN=   1.000    DCT BITS= 276

[illegible] DONE!
```

FIGURE 2.5-2 EXAMPLE OPERATION OF THE ATC PROGRAM

```
>RUN ATC70
SHALL WE DO SERIALIZATION OF DATA(Y/N)Y
SHALL WE INSEFT ERRORS IN CHANNEL(Y/N)?Y
ERROR RATE IN E11.4=
1.E-2
SHALL WE ENCCODE AND DECODE(Y/N)?Y
USE PITCH WEIGHTING(Y/N)? Y
USE FAST SORT(Y/N)? Y
SAMPLING RATE AND XMIT DATA RATE IN 2I6=6400,9600
IS THE INPUT ON MAG. TAPE? N
IS THE OUTPUT GOING TO MAG TAPE? N
OUTPUT FILE NAME= NL:
INPUT FILE NAME- SPEECH.DAT
NO FRAMES= 10
TOTAL NUMBER OF BITS=   369
INITIAL FRAME=5
FRAME   5   SN= 22.12   CSN= 22.12   PITCH=  1   P.GAIN=  1.000   DCT BITS= 276

FRAME=  6   SN= 13.77   CSN= 17.94   PITCH=  1   P.GAIN=  1.000   DCT BITS= 276

FRAME=  7   SN= 22.25   CSN= 19.38   PITCH=  1   P.GAIN=  1.000   DCT BITS= 276

FRAME=  8   SN= 16.51   CSN= 18.66   PITCH=  1   P.GAIN=  1.000   DCT BITS= 276

FRAME=  9   SN= 17.39   CSN= 18.41   PITCH=  1   P.GAIN=  1.000   DCT BITS= 276

FRAME= 10   SN= 16.05   CSN= 18.02   PITCH=  1   P.GAIN=  1.000   DCT BITS= 276

FRAME= 11   SN= 16.55   CSN= 17.81   PITCH=  1   P.GAIN=  1.000   DCT BITS= 276

FRAME= 12   SN= 19.24   CSN= 17.99   PITCH=  1   P.GAIN=  1.000   DCT BITS= 276

FRAME= 13   SN= 23.89   CSN= 18.64   PITCH=  1   P.GAIN=  1.000   DCT BITS= 276

FRAME= 14   SN= 23.21   CSN= 19.10   PITCH=  1   P.GAIN=  1.000   DCT BITS= 276

FRAME= 15   SN= 16.59   CSN= 18.87   PITCH=  1   P.GAIN=  1.000   DCT BITS= 276

ATC70 DONE!

>
```

FIGURE 2.5-3 EXAMPLE OPERATION OF THE ATC PROGRAM

2.6 Summary and Conclusions

The ATC optimization studies resulted in an ATC system which does not degrade significantly with a BER of 10^{-2} at a data rate of 9600 b/s. The specifications for the optimized system are shown in Table 2.6-1. The actual quantization tables can be found in either the FORTRAN listing of Appendix D or in tables within section 3 of Volume 2 describing the real-time MAP software.

The voice quality produced by the 9600 b/s ATC simulations is the best of any technique now known to GTE. The technique, however, is numerically complex requiring the complete processing capability of the CSP, Inc. MAP-300 floating point processor. Thus, for ATC to be practical, either higher speed hardware must be built or the technique must be simplified.

Future speech digitization development at 9600 cannot ignore the ATC algorithm because even though the technique is complex, it shows that good quality speech is possible at this data rate. Thus, the ATC technique developed under this contract will serve as a benchmark or standard to compare all new 9600 b/s speech digitization algorithms.

PARAMETER	SPECIFICATION
Input Bandwidth	0-3200 Hz
Sampling Rate	6400 Hz
Frame Rate	26.016/sec.
Number of Samples/Frame	246
Number of Samples Overlapped/Frame	10
Bits/Frame	369
Pitch	6 if voiced; 0 if unvoiced
Pitch Gain	2 if voiced; 0 if unvoiced
Voiced/Unvoiced	1
RMS Energy	5
DC BIAS	5
PARCOR 1	5
PARCOR 2	5
PARCOR 3	4
PARCOR 4	4
PARCOR 5	3
PARCOR 6	3
PARCOR 7	2
PARCOR 8	2
Parity Bits (Error Correction)	54
SYNC	1
DCT Coefficients	267 voiced; 275 unvoiced
Number of Error Control Blocks/Frame	3
Error Control Technique	(63,45) BCH

TABLE 2.6-1: OPTIMIZED ATC SYSTEM SPECIFICATION

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Appendix A Primitive BCH Codes

The BCH codes described in this appendix are cyclic codes that are well defined in terms of the roots of the generator polynomials [1]. These codes were discovered by Bose and Chaudhuri [2] - [3] and separately by Hocquenghem [4]. A binary (n,k) BCH code word consists of n symbols (bits in the binary case) where the first k bits are the information bits and the remaining $r = n-k$ bits are redundant parity checks. It is convenient to represent code words with polynomials as

$$f(x) = f_0 + f_1 x + \dots + f_{n-1} x^{n-1}, \quad f_i = 0 \text{ or } 1 \qquad \text{(A1)}$$

where each bit position is associated with a locator. If f(x) is a code word, then

$$f_1(x) = f_1 + f_2\, x + \dots + f_{n-1}\, x^{n-2} + f_0\, x^{n-1} \qquad \text{(A2)}$$

is also a codeword in a cyclic codes. In the primitive BCH code, which is the most convenient and powerful BCH code in theory and practice, the block length of the code may be defined as

$$n = 2^m - 1 \qquad \text{(A3)}$$

and with mt parity checks, it can correct any set of t independent errors within the block of n bits, where m and t are arbitrary positive integers [5]. This code may be described conveniently with the aid of finite Galois field theory introduced in Appendix B.

Let α be a primitive element of the finite field $GF(2^m)$, then the primitive BCH code may be described as the set of polynomials such that

$$f(\alpha^i) = 0,\ i = 1, 3, 5, \ldots, 2t - 1 \qquad (A4)$$

It is known in coding theory that these polynomials consist of all multiples of a single polynomial $g(x)$, known as the generator polynomial. This polynomial also satisfies the equations as

$$g(\alpha^i) = 0,\ i = 1, 3, 5, \ldots, 2t - 1 \qquad (A5)$$

These generator polynomials are tabulated in Table A for the selected primitive BCH codes.

Encoding Procedures

Let the k information bits be represented by the polynomial $d(x)$ as

$$d(x) = \sum_{i=0}^{k-1} d_i x^i \qquad (A6)$$

then, the code word of n bits may be expressed as

$$f(x) = x^{n-k} d(x) + r(x) \qquad (A7)$$

where $r(x)$ is the remainder (parity check) obtained according to the following equation:

$$\frac{x^{n-k} d(x)}{g(x)} = q(x) + \frac{r(x)}{g(x)} \qquad (A8)$$

Block length n	k	t	Generator Polynomial
63	57	1	$g_1(x) = (6, 1, 0) = x^6 + x + 1$
	51	2	$g_3(x) = g_1(x) \cdot (6, 4, 2, 1, 0)$
	45	3	$g_5(x) = g_3(x) \cdot (6, 4, 2, 1, 0)$
127	120	1	$g_1(x) = (7, 3, 0) = x^7 + x^3 + 1$
	113	2	$g_3(x) = g_1(x) \cdot (7, 3, 2, 1, 0)$
	106	3	$g_5(x) = g_3(x) \cdot (7, 4, 3, 2, 0)$
255	247	1	$g_1(x) = (8, 4, 3, 2, 0) = x^8 + x^4 + x^3 + x^2 + 1$
	239	2	$g_3(x) = g_1(x) \cdot (8, 6, 5, 4, 2, 1, 0)$
	231	3	$g_5(x) = g_3(x) \cdot (8, 7, 6, 5, 4, 2, 0)$

TABLE A: GENERATOR POLYNOMIALS FOR SELECTED PRIMITIVE BCH CODES where g(x) is the generator polynomial of the code. Therefore, encoding can be performed by the following procedures:

1). Calculate x^{n-k} d(x) by left shifting the information bits n-k times

2). Calculate the remainder (parity bits) r(x) from the division of x^{n-k} d(x) by g(x)

3). Add the polynomial x^{n-k} d(x) and r(x) to form the code word

The procedures of 1) and 3) can be done simply by shifting and addition. However, the procedure of 2) is rather involved in computation if the actual division is performed to get the remainder. If the BCH code is specified and it is desired to speed up the processing time of 2), it is recommended to use a look-up table procedure for the calculation of the remainder from 2). The code word is then transmitted through the noisy channel, where the received code word may be altered depending on the introduction of channel errors.

Decoding Procedures

There are several algorithms for a decoding of BCH codes. Efficient decoding algorithms have been discovered for BCH codes [1] - [7]. The Berlekamp decoder is particularly attractive for powerful codes that provide for a good deal of error corrections (e.g., 10 or more). The Peterson algorithm, however, is more efficient for less powerful codes (e.g., the codes used in generalized burst trapping). In this decoding procedure, the problem of finding efficient solutions to the key decoding equation will be addressed by using the Peterson technique.

When a BCH code word $\{f(x)\}$ is transmitted over a noisy channel, this code word may be corrupted by the channel, and what is received $\{\gamma(x)\}$ can be different from the intended code word. Thus, the received word may be expressed as

$$\gamma(x) = f(x) + e(x) \tag{A9}$$

where $e(x)$ is the error polynomial which a decoder must compute to correct errors introduced by the channel. Let the received data be expressed in vector γ as

$$\gamma = [\gamma_0, \gamma_1, \ldots, \gamma_{n-1}] \tag{A10}$$

or its associated polynomial $\gamma(x)$ by

$$\gamma(x) = \gamma_0 + \gamma_1 x + \ldots + \gamma_{n-1} x^{n-1} \tag{A11}$$

Denote each of the error location numbers by β_j, $j = 1, 2, \ldots, t$, then it is shown [1] that the power sums S_i can be expressed as

$$\begin{aligned} S_i &= \gamma(\alpha^i) \\ &= \sum_{j=1}^{t} \beta_j^i , \qquad i = 1, 3, 5, \ldots, 2t-1 \end{aligned} \tag{A12}$$

In order to find the error locations, the Peterson procedures consist of three steps:

Step 1: Compute the power sums S_i from the received sequence through the relations

$$S_i = \gamma(\alpha^i), \ i = 1, 3, 5, \ldots, 2t-1$$

$$S_{2i} = S_i^2 \tag{A13}$$

Step 2: Compute the symmetric functions σ_k, $k = 1, 2, \ldots, t$ from the power sums S_i, i.e.,

$$\sigma(x) = x^t + \sigma_1 x^{t-1} + \ldots + \sigma_{t-1} x + \sigma_t$$

$$= (x + \beta_1)(x + \beta_2) \ldots (x + \beta_t) \tag{A14}$$

and the σ_k's may be obtained by the use of Newton's identities [1]

$$\underline{\sigma} = \begin{bmatrix} \sigma_1 \\ \sigma_2 \\ \cdot \\ \cdot \\ \cdot \\ \sigma_t \end{bmatrix}$$

$$= M_t^{-1} \underline{S} \tag{A15}$$

$$= \begin{bmatrix} 1 & 0 & 0 & 0 \ldots & 0 \\ S_2 & S_1 & 1 & 0 \ldots & 0 \\ \cdots & & & & \\ \cdots & & & & \\ S_{2t-2} & S_{2t-3} & S_{2t-4} & \cdots & S_{t-1} \end{bmatrix}^{-1} \begin{bmatrix} S_1 \\ S_3 \\ \cdot \\ \cdot \\ \cdot \\ S_{2t-1} \end{bmatrix}$$

If the determinant of M_t is singular, then reduce the error number t by 2 and proceed with it again.

Step 3: Find the error position locator β_j, $j = 1, 2, \ldots, t$, which is the roots of the polynomial $\sigma(x)$ in eq. (A14).

An efficient algorithm for calculating the β_j's from eq. (A14) has been developed by Chien [5], and all that remains to completely specify a binary BCH decoder is the computation of the coefficients of error locator polynomial, σ_j's. As it is noted from eq. (A15), the calculation of the σ_j's involved matrix inversion which can be expressed analytically for the case $t \leq 3$. The results are:

For t = 1,

$$\sigma_1 = S_1$$

For t = 2,

$$\sigma_1 = S_1$$
$$\sigma_2 = (S_3 + S_1^3)/S_1$$

For t = 3,

$$\sigma_1 \quad S_1$$
$$\sigma_2 = (S_1^2 S_3 + S_5)/(S_1^3 + S_3)$$
$$\sigma_3 = (S_1^3 + S_3) + S_1\sigma_2$$

The calculation of the σ_i's and the estimation of the error number are shown in Figure A1 for t = 3. The flowchart of Chien's search decoding

$$m_1(\alpha) = \alpha^6 + \alpha + 1$$
$$= 0$$
$$= \alpha^{63} + 1$$

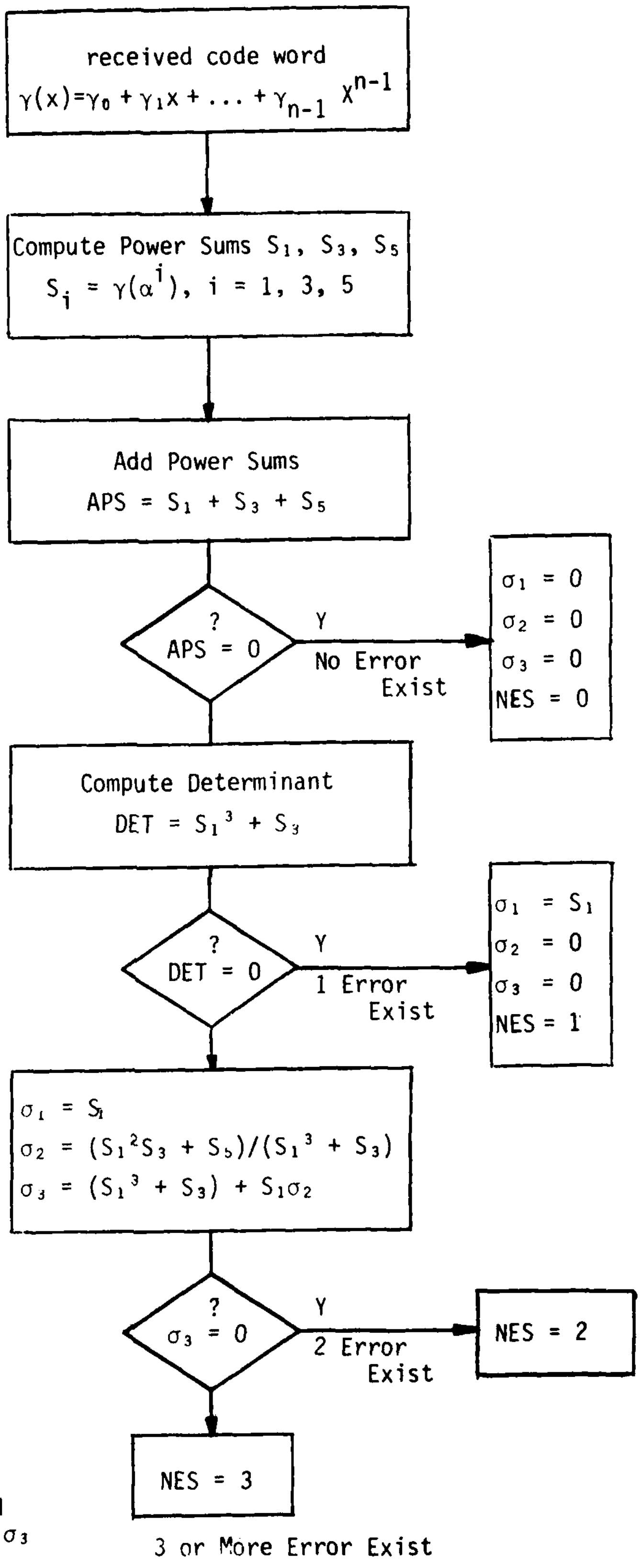

FIGURE A1: COMPUTATION OF σ_1, σ_2, σ_3

procedure is shown in Figure A2. This flowchart is for $t = 3$, i.e., the decoding algorithm can correct errors up to 3. One interesting observation in this decoding procedure is that the correction of errors may be performed erroneously if the number of errors in the block is greater than 3. Hence, the corrections may introduce additional channel errors. In order to avoid these additional errors, error corrections are made only when the estimated error number (NES in Figure A1) equals to the measured error number (K in Figure A2). This procedure eliminates most of the additional errors when more than 3 errors exist in the received word. In other words, the detection of errors more than 3 (i.e., 4, 5, 6, ..., etc.) is feasible most of the cases. This fact contributes some improvements of the coder performance when the channel is very noisy (bit error rate $\approx 10^{-2}$).

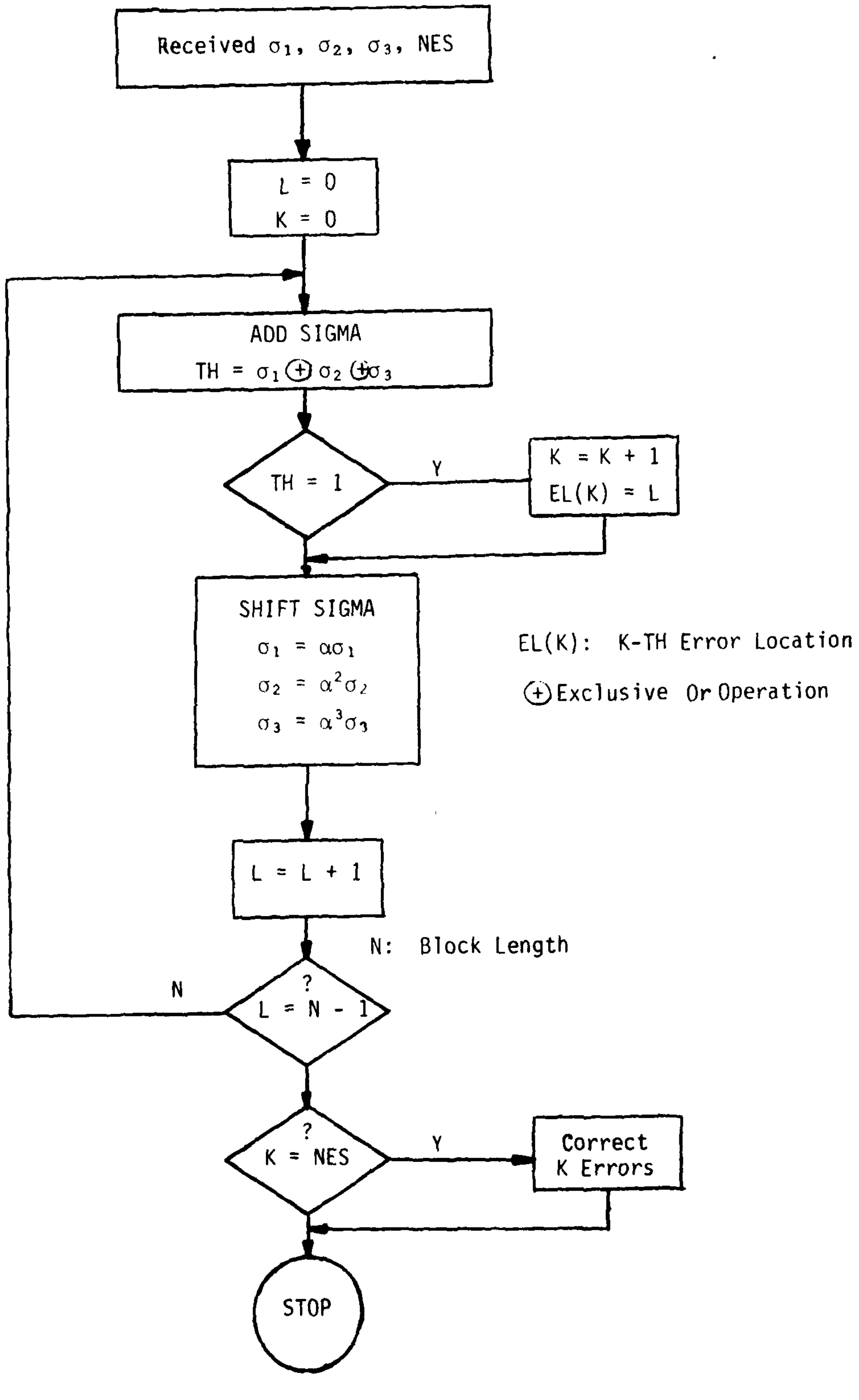
Received σ_1, σ_2, σ_3, NES
L = 0
K = 0
ADD SIGMA
TH = $\sigma_1 \oplus \sigma_2 \oplus \sigma_3$
TH = 1
Y
K = K + 1
EL(K) = L
SHIFT SIGMA
$\sigma_1 = \alpha\sigma_1$
$\sigma_2 = \alpha^2\sigma_2$
$\sigma_3 = \alpha^3\sigma_3$
EL(K): K-TH Error Location
$\oplus$ Exclusive Or Operation
L = L + 1
N: Block Length
N
$L \overset{?}{=} N - 1$
$K \overset{?}{=} NES$
Y
Correct K Errors
STOP

FIGURE A2: CHIEN'S SEARCH DECODING PROCEDURE

References for Appendix A

[1] Peterson, W. W., Error-Correcting Codes, The MIT Press, Cambridge, MA, 1961.

[2] Bose, R. C. and D. K. Ray-Chandhuri, "On a Class of Error-Correcting Binary Group Codes," IEEE Trans. on Information and Control, Vol. IC-3, pp. 68-79, 1960.

[3] Bose, R. C. and D. K. Ray-Chandhuri, "Further Results on Error-Correcting Binary Group Codes," IEEE Trans. on Information and Control, Vol. IC-3, pp. 279-290, 1960.

[4] Hocquenghem, A., "Codes Correcteurs d'erreurs," Chiffres, Vol. 2, pp. 147-156, 1959.

[5] R. T. Chien, "Cyclis Decoding Procedures for Bose-Chandhuri-Hocquenghem Codes," IEEE Trans. on Information Theory, Vol. IT-10, pp. 357-363, 1964.

[6] E. R. Berlekamp, Algebraic Coding Theory, New York, McGraw-Hill, 1968.

[7] J. L. Massey, "Shift-Register Synthesis and BCH Decoding," IEEE Trans. on Information Theory, Vol. IT-15, No. 1, pp. 122-127, 1969.

Appendix B Operations in Galois Field

A Galois field is a finite set of elements that satisfy the axioms of a general field. Two operations (addition and multiplication) and their inverses are defined on the field elements. There is an identity element for each field element for both of the operations (0, 1) that is itself in the field. Also, both addition inverses and multiplication inverses are in the field. Finally, the rules of commutation and associativity are obeyed by the elements of the field.

Consider the following sixteen polynomials and their vector binary representations.

0	0000
1	0001
$1 + X$	0011
$1 + X + X^2$	0111
$1 + X + X^2 + X^3$	1111
X	0010
$X + X^2$	0110
$X + X^2 + X^3$	1110
X^2	0100
$X^2 + X^3$	1100
X^3	1000
$1 + X^3$	1001
$1 + X^2$	0101
$1 + X^2 + X^3$	1101
$X + X^3$	1010
$1 + X + X^3$	1011

As long as addition and multiplication of these polynomials is defined so that the axioms for the field are obeyed, then this will, in fact, be a Galois field of 2^4 elements (GF(2^4)).

Addition is defined to be modulo 2. Each element is its own additive inverse and addition and subtraction of elements are the same.

Multiplication must be defined so that the product of two elements does not take us out of the field. For this reason, multiplication in a Galois field is not ordinary multiplication of polynomials. Rather, multiplication is defined modulo an irreducible polynomial, the primitive polynomial of the Galois field. For our field GF(2^m), the primitive polynomial is $1 + X + X^4$. To generate the 16 vectors in the field, all one needs to do is to divide X^m where m = 0, 1, ... 14 by the primitive polynomial.

0	-1
1	0
X	X
X^2	X^2
X^3	X^3
X^4	$1 + X$
X^5	$X + X^2$
X^6	$X^2 + X^3$
X^7	$X^3 + X + 1$
X^8	$X^2 + 1$
X^9	$X^3 + X$
X^{10}	$X^2 + x + 1$
X^{111}	$X^3 + X^2 + 1$
X^{12}	$X^3 + X^2 + X + 1$
X^{13}	$X^3 + X^2 + 1$
X^{14}	$X^3 + 1$
X^{15}	$X^0 + 1$

$$\begin{array}{r|l} & 1 \qquad R[1 + X] \\ \hline 1 + X + X^4 & X^4 \end{array}$$

$$\begin{array}{r|l} & X \qquad R[X + X^2] \\ \hline 1 + X + X^4 & X + X^5 \end{array}$$

It is now seen that the product of two binary vectors in the field is just the sum of their powers. The table repeats every fifteen powers so it is all done modulo 15.

$$X^{i} + X^{j} = X^{i+j} \text{ (mod 15)}$$

$$\frac{X^{i}}{X^{j}} = X^{i-j} \text{ (mod 15)}$$

APPENDIX C

FORTRAN Source Listings for the ATC Simulation

This appendix contains the FORTRAN source programs for the ATC simulation. The first page of this listing is a compile file which uses FORTRAN-IV PLUS to generate object files from the source files and which sends listings to the line printer. The second page of these programs is the overlay description language (ODL) needed to build the ATC task under the RSX-11M operating system. The remainder of the appendix includes the main program and subroutines. The order of the programs follows the order of the files as listed in the overlay description language on the second page.

```
PIP ATCOLA.CMD;*/PU
PIP BUILD.CMD;*/PU
PIP ATC70.ODL;*/PU
PIP ATC70.OBJ;*/DE
PIP OVR1.OBJ;*/DE
PIP OVR2.OBJ;*/DE
PIP TAPE3.OBJ;*/DE
PIP OVR3.OBJ;*/DE
PIP OVR4.OBJ;*/DE
PIP OVR5.OBJ;*/DE
PIP OVR6.OBJ;*/DE
PIP OVR7.OBJ;*/DE
PIP OVR8.OBJ;*/DE
PIP OVR9.OBJ;*/DE
PIP OVR10.OBJ;*/DE
PIP OVR11.OBJ;*/DE
PIP OVR12.OBJ;*/DE
PIP OVR13.OBJ;*/DE
PIP OVEVOD.OBJ;*/DE
PIP FASTF.OBJ;*/DE
PIP GF2POL.OBJ;*/DE
PIP GENTAB.OBJ;*/DE
PIP LOOKUP.OBJ;*/DE
PIP INVLOK.OBJ;*/DE
PIP CHARTS.OBJ;*/DE
PIP GF2DIV.OBJ;*/DE
PIP GF2MUL.OBJ;*/DE
PIP GF2ADD.OBJ;*/DE
PIP ATCOLA.CMD/SP
PIP BUILD.CMD/SP
PIP ATC70.ODL/SP
F4P ATC70,ATC70=ATC70/DE
F4P TAPE3,TAPE3=TAPE3
F4P OVEVOD,OVEVOD=OVEVOD
F4P FASTF,FASTF=FASTF
F4P OVR1,OVR1=OVR1
F4P OVR2,OVR2=OVR2
F4P OVR3,OVR3=OVR3
F4P OVR4,OVR4=OVR4
F4P OVR5,OVR5=OVR5
F4P OVR6,OVR6=OVR6
F4P OVR7,OVR7=OVR7
F4P OVR8,OVR8=OVR8
F4P OVR9,OVR9=OVR9
F4P OVR10,OVR10=OVR10
F4P GF2POL,GF2POL=GF2POL
F4P GENTAB,GENTAB=GENTAB
F4P LOOKUP,LOOKUP=LOOKUP
F4P INVLOK,INVLOK=INVLOK
F4P CHARTS,CHARTS=CHARTS
F4P GF2DIV,GF2DIV=GF2DIV
F4P GF2MUL,GF2MUL=GF2MUL
F4P GF2ADD,GF2ADD=GF2ADD
F4P OVR11,OVR11=OVR11
F4P OVR12,OVR12=OVR12
F4P OVR13,OVR13=OVR13
```

```
             .NAME CNTRL[illegible]
        .[illegible] A-[illegible](E,[illegible] [illegible] G,H,I,J,K,L,M,N),Z
               [illegible]

[illegible]      .[illegible]
[illegible]      .FCTR OVR3
[illegible]      .FCTR [illegible]
[illegible]      .FCTR [illegible]
[illegible]      .FCTR [illegible]
[illegible]      .FCTR OVR[illegible]
I:      .FCTR OVR8
J:      .FCTR OVR9
K:      .FCTR OVR10-GF2POL-LOOKUP-INVLOK-CHARTS-*(GENTAB)
L:      .FCTR OVR11
M:      .FCTR OVR12
N:      .FCTR OVR13
Z:      .FCTR CNTRL1-*(AA,GF2ADD,BB,TAPE3)
AA:     .FCTR OVEVOD-*(FASTF)
BB:     .FCTR GF2MUL-*(GF2DIV)
        .END
```

```
C     PROGRAM NAME:ATC70.FTN  ORIGINATED:02-DEC-77
C                                    UPDATE:13-SEP-79
C     MAP-300 BENCHMARK PROGRAM
C     QUANTIZATION REQUIREMENTS:
C     DCT PARAMETERS: PROGRAMMABLE
C     PARCORS: 28 BITS/FRAME
C     DC BIAS: 4 BITS/FRAME
C     VARIANCE: 5 BITS/FRAME
C     PITCH: 7 BITS/FRAME
C     PITCH GAIN: 2 BITS/FRAME
C     SYNCRONIZATION: 1 BIT/FRAME
C
C
C
      COMMON/MTAPE0/NIN(256),NOUT(256)
      COMMON/MTAPE1/NSKIP,IST,NTCTI,NTUPS,NTOTO
      COMMON/MTAPE2/NEND,NERR,NFILE,NINS,NOUTS
      COMMON/MTAPE3/NBF(1324),NBUF(1324)
      COMMON/MTAPE4/LST,IBEG
      COMMON/MTAPE5/MASK,ISW(2),IOATT,IOSUC,IEALN,IORWD
     1,IOWLB,IVER,IOSPF,IEEOF,IOEOF,IORLB,MT0(6),MT1(6),DSW
      COMMON/MTAPE6/APPEND
C
      COMMON/SORT/DCT1(256),DCT2(256),IORDR(256),XR(512),XI(512)
      COMMON/DITBA/QTDCT(256),QTPAR(8),QTDC,QTVAR,QTM,QTG
      COMMON/DECBCH/NES,NDB
      COMMON/BLOCK1/INIT1,DCT(256),LTH,NSTAGE
      COMMON/BLOCK2/A(8),R(11),U,PARCOR(3)
      COMMON/BLOCK3/DTPAR(8),DTA(8),LPCN
      EQUIVALENCE (Y,PWEIT)
      EQUIVALENCE  (DCT,DRDCT),(DTPAR,DRPAR),(DTA,DRA)
      COMMON/BLOCK4/LTH2
      COMMON/BLOCK5/LTH3,LTH4,NP,XN,LP1,ARG
     1,DLOG2,CNS,ICOUNT,IREC,NBPF,NBRPF
     2,PI,NSTAG,BTLTH,PWATE,TYPSRT,NEPB
      COMMON/BLOCK6/IBIT(256),IPDEC,INBA(500)
      COMMON/SW/ICOUN3,IPRSW
      INTEGER QTDCT,QTPAR,QTDC,QTVAR,QTM,QTG
      INTEGER QRDCT(256),QRPAR(8),QRDC,QRVAR,QRM,QRG
      EQUIVALENCE (QTDCT,QRDCT),(QTPAR,QRPAR)
      INTEGER BITSP(8)
      LOGICAL*1 ERRDEC,SER,ENCDEC,PWATE,YES,NO,TYPSRT
C     ALTERED INPUT AND OUTPUT
      DIMENSION X(256),YY(10),Y(256)
C     DCT COMPONENTS
      DIMENSION DRDCT(256)
C     PITCH WEIGHTING FUNCTION
      DIMENSION PWEIT(256)
C     ERROR INSERTION ROUTINE CEIR
      DIMENSION NERB(6)
C     QUANTIZATION PARAMETERS
      DIMENSION DCTHR(16),DCDEC(16)
      DIMENSION DCTTHR(96),DCTDEC(96)
      DIMENSION VARTHR(32),VARDEC(32)
      DIMENSION PGTHR(4),PGDEC(4)
      DIMENSION PTHR(256),PDEC(256)
      DIMENSION DRPAR(8),DRA(8)
      DATA YES/'Y'/
      DATA NO/'N'/
C     PARCOR BIT ALLOCATION
      DATA BITSP/5,5,4,4,3,3,2,2/
C     PARCOR(I) QUANTIZER THRESHOLDS
      DATA PTHR/
```

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SPEECH OPTIMIZATION AT 9600 BITS/SECOND. VOLUME 1. SOFTWARE SIM--ETC(U)
SEP 80 A J GOLDBERG, L COSELL, S KWON DCA100-78-C-0064

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END
DATE
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```
     1 0.16391E-01,0.33470E-01,0.51313E-01,0.69970E-01,
     1 0.89376E-01,0.11219E+00,0.13700E+00,0.16288E+00,
     1 0.18973E+00,0.21909E+00,0.25131E+00,0.28606E+00,
     1 0.32420E+00,0.36803E+00,0.41824E+00,0.47689E+00,
     1 0.54314E+00,0.61470E+00,0.69616E+00,0.77133E+00,
     1 0.84345E+00,0.92970E+00,0.10190E+01,0.11218E+01,
     1 0.12403E+01,0.13426E+01,0.14437E+01,0.15414E+01,
     1 0.16443E+01,0.17886E+01,0.19332E+01,0.1E+21,
     1 0.17514E+00,0.31624E+00,0.41542E+00,0.50895E+00,
     1 0.59019E+00,0.65272E+00,0.70686E+00,0.76395E+00,
     1 0.81878E+00,0.87625E+00,0.94152E+00,0.10123E+01,
     1 0.10847E+01,0.11620E+01,0.12381E+01,0.13059E+01,
     1 0.13727E+01,0.14357E+01,0.14920E+01,0.15468E+01,
     1 0.16017E+01,0.16527E+01,0.16951E+01,0.17323E+01,
     1 0.17683E+01,0.18035E+01,0.18365E+01,0.18666E+01,
     1 0.18961E+01,0.19267E+01,0.19616E+01,0.1E+21,
     1 0.31528E+00,0.45224E+00,0.56230E+00,0.65753E+00,
     1 0.74494E+00,0.82867E+00,0.91130E+00,0.99458E+00,
     1 0.10795E+01,0.11678E+01,0.12608E+01,0.13581E+01,
     1 0.14619E+01,0.15854E+01,0.17491E+01,0.1E+21,16*0.0,
     1 0.40927E+00,0.57217E+00,0.69580E+00,0.80120E+00,
     1 0.89689E+00,0.98763E+00,0.10764E+01,0.11650E+01,
     1 0.12547E+01,0.13464E+01,0.14408E+01,0.15388E+01,
     1 0.16420E+01,0.17540E+01,0.18815E+01,0.1E+21,16*0.0,
     1 0.55688E+00,0.74670E+00,0.89149E+00,0.10220E+01,
     1 0.11526E+01,0.12967E+01,0.14823E+01,0.1E+21,24*0.0,
     1 0.57331E+00,0.76317E+00,0.90902E+00,0.10407E+01,
     1 0.11725E+01,0.13190E+01,0.15093E+01,0.1E+21,24*0.0,
     1 0.56750E+00,0.87517E+00,0.11666E+01,0.1E+21,28*0.0,
     1 0.72199E+00,0.98497E+00,0.12430E+01,0.1E+21,28*0.0/
C      PARCOR(J) DEQUANTIZER DECISIONS
      DATA PDEC/
     1 0.80431E-02,0.24740E-01,0.42199E-01,0.60427E-01,
     1 0.79512E-01,0.10024E+00,0.12414E+00,0.14986E+00,
     1 0.17589E+00,0.20368E+00,0.23451E+00,0.26811E+00,
     1 0.30401E+00,0.34440E+00,0.39167E+00,0.44480E+00,
     1 0.50898E+00,0.57740E+00,0.65201E+00,0.74030E+00,
     1 0.80236E+00,0.88455E+00,0.97486E+00,0.10631E+01,
     1 0.11805E+01,0.13001E+01,0.13851E+01,0.15023E+01,
     1 0.15805E+01,0.17030E+01,0.18691E+01,0.19972E+01,
     1 0.87097E-01,0.26319E+00,0.36928E+00,0.46156E+00,
     1 0.55635E+00,0.62403E+00,0.68141E+00,0.73631E+00,
     1 0.79160E+00,0.84597E+00,0.90653E+00,0.97650E+00,
     1 0.10480E+01,0.11214E+01,0.12027E+01,0.12736E+01,
     1 0.13383E+01,0.14072E+01,0.14643E+01,0.15197E+01,
     1 0.15740E+01,0.16295E+01,0.16759E+01,0.17142E+01,
     1 0.17504E+01,0.17863E+01,0.18208E+01,0.18521E+01,
     1 0.18814E+01,0.19108E+01,0.19426E+01,0.19807E+01,
     1 0.23813E+00,0.39243E+00,0.51204E+00,0.61256E+00,
     1 0.70251E+00,0.78737E+00,0.86998E+00,0.95262E+00,
     1 0.10365E+01,0.11226E+01,0.12129E+01,0.13086E+01,
     1 0.14075E+01,0.15162E+01,0.16545E+01,0.18438E+01,16*0.0,
     1 0.31411E+00,0.50442E+00,0.63992E+00,0.75167E+00,
     1 0.85074E+00,0.94305E+00,0.10322E+01,0.11205E+01,
     1 0.12094E+01,0.13000E+01,0.13929E+01,0.14888E+01,
     1 0.15880E+01,0.16951E+01,0.18129E+01,0.19500E+01,16*0.0,
     1 0.44510E+00,0.66865E+00,0.82476E+00,0.95821E+00,
     1 0.10859E+01,0.12194E+01,0.13739E+01,0.15907E+01,24*0.0,
     1 0.46200E+00,0.68463E+00,0.84172E+00,0.97631E+00,
     1 0.11050E+01,0.12400E+01,0.13980E+01,0.16206E+01,24*0.0,
     1 0.39739E+00,0.73761E+00,0.10127E+01,0.13204E+01,28*0.0,
     1 0.57856E+00,0.86541E+00,0.11045E+01,0.13815E+01,28*0.0/
      DATA DCTHR/
     1 0.12475E+00,0.25019E+00,0.37706E+00,0.50625E+00,
     1 0.63881E+00,0.77594E+00,0.91903E+00,0.10699E+01,
     1 0.12309E+01,0.14053E+01,0.15977E+01,0.18155E+01,
     1 0.29718E+01,0.23938E+01,0.28541E+01,0.10000E+21/
      DATA DCDEC/
     1 0.62293E-01,0.18721E+00,0.31317E+00,0.44096E+00,
     1 0.57154E+00,0.70609E+00,0.84579E+00,0.99227E+00,
     1 0.11475E+01,0.13143E+01,0.14964E+01,0.16991E+01,
     1 0.19319E+01,0.22117E+01,0.25759E+01,0.31324E+01/
      DATA VARTHR/
     1  0.56312E+01,0.77983E+01,0.98311E+01,0.12057E+02,
     1  0.14548E+02,0.16994F+02,0.19276E+02,0.21426E+02,
     1  0.23405E+02,0.25415E+02,0.27476E+02,0.29409E+02,
     1  0.31179E+02,0.32888E+02,0.34633E+02,0.36477E+02,
     1  0.38386E+02,0.40204E+02,0.41864E+02,0.43443E+02,
     1  0.44982E+02,0.46483E+02,0.47960E+02,0.49431E+02,
     1  0.50900E+02,0.52334E+02,0.53709E+02,0.55072E+02,
     1  0.56491E+02,0.58016E+02,0.59760E+02,0.10900E+21/
      DATA VARDEC/
     1 0.44600E+01,0.68023E+01,0.87943E+01,0.10868E+02,
     1 0.13247E+02,0.15849E+02,0.18140E+02,0.20412E+02,
     1 0.22439E+02,0.24372E+02,0.26458E+02,0.28495E+02,
     1 0.30322E+02,0.32035E+02,0.33741E+02,0.35525E+02,
     1 0.37430E+02,0.39343E+02,0.41065E+02,0.42664E+02,
     1 0.44222E+02,0.45741E+02,0.47224E+02,0.48695E+02,
     1 0.50167E+02,0.51633E+02,0.53034E+02,0.54385E+02,
     1 0.55758E+02,0.57223E+02,0.58810E+02,0.60709E+02/
      DATA PGTHR/
     1 0.48797E+00,0.66711E+00,0.82399E+00,0.10000E+21/
      DATA PGDEC/
     1 0.39165E+00,0.58430E+00,0.74993E+00,0.89805E+00/
      DATA DCTTHR/
     1 16*1.0E+21,
     1 1.0E+21,15*0.0,
     1 1.1269,1.0E+21,14*0.0,
     1 0.5332,1.2527,2.3796,1.0E+21,12*0.0,
     1 0.2644,0.5667,0.9198,1.3444,1.8776,2.5971,3.7240,
     1 1.0E+21,8*0.0,
     1 0.1322,0.2732,0.4243,0.5870,0.7632,0.9555,1.1669,
     1 1.4019,1.6663,1.9687,2.3217,2.7463,3.2795,3.9990,
     1 5.1259,1.0E+21/
      DATA DCTDEC/
     1 16*0.0,
     1 0.7071,15*0.0,
     1 0.4196,1.8340,14*0.0,
     1 0.2334,0.3330,1.6725,3.0867,12*0.0,
     1 0.1240,0.4048,0.7287,1.1110,1.5778,2.1773,3.0169,
     1 4.4311,8*0.0,
     1 0.0640,0.2004,0.3461,0.5025,0.6715,0.8550,1.0559,
     1 1.2779,1.5259,1.8068,2.1306,2.5129,2.9797,3.5793,
     1 4.4188,5.8330/
C
C     INITIALIZE ENTIRE SYSTEM
C
100   FORMAT(I3)
C     <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
C     FIRST SET LPC ORDER TO 8
      LPCH=8
C     NOW SET THE VOICING DECISION THREWSHOLD TO BE 35
C     IF THE RMS IS GREATER THAN 35 THAN IT IS VOICED,
C     IF THE RMS IS LESS THAN 35 THAN IT IS UNVOICED
C     AND THEREFORE NOT PITCH WEIGHTED.
      IPTHR=35
C     <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
C     NOW QUANTIZATION OF DECIMAL DATA IS DECIDED.
C     DECODING AND ENCODING. AND CHANNEL ERROR INSERTION RATE
C     FOR LINE SIMULATION
```

```
C
      WRITE(5,7302)
7302  FORMAT(1H$,'SHALL WE DO SERIALIZATION OF DATA(Y/N)')
      READ(5,2601,END=9999,ERR=9999)SER
2601  FORMAT(A1)
      IF(SER.EQ.NO)GO TO 7307
      WRITE(5,7313)
7313  FORMAT(1H$,'SHALL WE INSERT ERRORS IN CHANNEL(Y/N)?')
      READ(5,2602,END=9999,ERR=9999)ERRDEC
      IF(ERRDEC.EQ.NO)GO TO 7303
      WRITE(5,7300)
7300  FORMAT(1X,'ERROR RATE IN E11.4=')
      READ(5,7301)PREA
7303  CONTINUE
      WRITE(5,7304)
7304  FORMAT(1H$,'SHALL WE ENCCODE AND DECODE(Y/N)?')
      READ(5,2602,END=9999,ERR=9999)ENCDEC
2602  FORMAT(A1)
7301  FORMAT(E11.4)
C
C     <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
C     OVR2 WILL SET UP FRAME CONSTANTS, SAMPLING RATE AND BIT ALLOCATIONS
7307  CALL OVR2
C
C     <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
C     BEGIN ANALYSIS/SYNTHESIS
C
C
      TYPE 808,NBRPF
      BTLTH2=BTLTH
808   FORMAT(1X,'TOTAL NUMBER OF BITS=',2X,I4)
      LP1=LPCN+1
      IVARF=0
      ICSUB=0
      ICOUNT=0
      DO 1002 I=1,NP
1002  YY(I)=0.0
      WRITE(5,2005)
2005  FORMAT(1H$,'INITIAL FRAME=')
      READ(5,1101)INITFR
1101  FORMAT(I6)
C
C     NOW REWIND THE TAPE
C
      IST=1
      NSKIP=1
      CALL TAPE3(6)
C
C
C     INITIALIZE RANDOM NUMBER GENERATOR
      IRN=0
      JRN=0
      DO 1003 I=1,49
1003  CALL RANDU(IRN,JRN,YOR)
C
C
1001  CONTINUE
      BTLTH=BTLTH2
      ICOUNT=ICOUNT+1
C     IF(ICOUNT.LT.16)GO TO 67
C     CALL CLOSE(4)
C     CALL ASSIGN(4,'GD0:')
C     PAUSE 'CHANGE IN LUN4 IS MADE,"RES"UME'
67    ICOUN3=ICOUNT
```

```
D          WRITE(4,899)ICOUNT
D899       FORMAT(1H1,'FRAME=',I6)
           IF(ICOUNT.GT.(IREC+INITFR)) GOTO 5000
C
C          <<<<<<<<<<<<<<<<<<<<<<
C          >>>>> TRANSMITTEP <<<<<
C          >>>>>>>>>>>>>>>>>>>>>>
C
C          DATA INPUT
C
           CALL TAPE3(1)
           IF(ICOUNT.LT.INITFR)GO TO 1001
D          WRITE(4,898)(I,NIN(I),I=1,LTH)
D898       FORMAT(/1X,4(1X,'NIN(',I3,')=',I6,2X))
           IF(NEND.NE.0) GO TO 5000
C
C          NORMALIZE BY DC BIAS & VARIANCE
C
           IPDEC=1
           DCBIAS=0.
           DO 280 I=1,LTH
280        DCBIAS=DCBIAS+NIN(I)
           DCBIAS=DCBIAS/LTH
           DO 281 I=1,LTH
281        X(I)=NIN(I)-DCBIAS
           VAR=0.
           DO 3002 I=1,LTH
3002       VAR=VAR+X(I)*X(I)
           VAR=SQRT(VAR/LTH)
           IF(VAR.LT.FLOAT(IPTHR))IPDEC=0
C          WRITE(5,3003)VAR
C3003      FORMAT(1X,'RMS=',3X,F15.8)
           IF(VAR.GT.0.)GO TO 3013
           IVARF=IVARF+1
           GO TO 1001
3013       CONTINUE
           DO 2 I=1,LTH
2          X(I)=X(I)/VAR
D          WRITE(4,900)DCBIAS
D          WRITE(4,901)VAR
C
C          COMPRESS STATISTICS
C
           DCBIAS=DCBIAS/VAR
           VAR=20.0*ALOG10(VAR)
D          WRITE(4,900)DCBIAS
D900       FORMAT(/1X,'DCBIAS=',F12.5)
D          WRITE(4,901)VAR
D901       FORMAT(/1X,'VAR=',F12.5)
C
C
C
C          DO DCT WITH ZERO PACKING TO AVOID CIRCULAR CONVOLUTION
C
           DO 21 I=1,LTH
           XR(I)=X(I)
           XR(I+LTH)=0.0
           XI(I)=0.0
21         XI(I+LTH)=0.0
           INIT1=1
           CALL OVP1
           CALL DCTSUB(LTH,XR,XI,DCT,1)
C          WRITE(4,920)(I,XR(I),I=1,2*LTH)
C          WRITE(4,921)(I,XI(I),I=1,2*LTH)
C          WRITE(4,923)(I,SIN(ARG*(I-1)),I=1,LTH)
```

```
D     WRITE(4,922)(I,COS(ARG*(I-1)),I=1,LTH)
D920  FORMAT(/1X,4(1X,'XR(',I3,')=',E15.8,2X))
D921  FORMAT(/1X,4(1X,'XI(',I3,')=',E15.8,2X))
D922  FORMAT(/1X,4(1X,'COS(',I3,')=',E15.8,2X))
D923  FORMAT(/1X,4(1X,'SIN(',I3,')=',E15.8,2X))
D     WRITE(4,902)(I,DCT(I),I=1,LTH)
D902  FORMAT(/1X,4(1X,'DCT(',I3,')=',E15.8,2X))
C
C     COMPUTE PSEUDO AUTOCORRELATION FUNCTION
C
C
C     FIRST DO EVEN ODD REFLECTION AROUND SYMMETRICSIGNAL
C     AND SCALE BY 2
C
      DCT1(1)=XR(1)/2.0
      DCT2(1)=0.0
      XI(1)=XR(2)/2.0
      XI(LTH+1)=XR(LTH2)/2.0
      MLTH2=LTH2+2
      DO 90 I=2,LTH
      J=2*I-1
      DCT1(I)=XR(J)/2.0
      DCT2(I)=XR(MLTH2-J)/2.0
      XI(I)=XR(J+1)/2.0
90    XI(LTH+I)=XR(MLTH2-J-1)/2.0
      DO 91 I=1,LTH
      XR(I+LTH)=DCT2(I)
91    XR(I)=DCT1(I)
C     NOTE DCT1 AND DCT2 ARE JUST TEMPORARY SHUFFLE VECTORS
D     WRITE(4,920)(I,XR(I),I=1,2*LTH)
D     WRITE(4,921)(I,XI(I),I=1,2*LTH)
C     CALL OVEVOD(LTH2,2,2)
C     CALL EVODD(LTH2,XR,XI,2,2)
D     WRITE(4,920)(I,XR(I),I=1,LTH)
D     WRITE(4,921)(I,XI(I),I=1,LTH)
C     SAVE PSEUDOCORRELATION
C
      DO 625 I=1,LP1
625   R(I)=XR(I)
D     WRITE(4,903)(I,R(I),I=1,LP1)
D903  FORMAT(/1X,5(1X,'R(',I2,')=',E15.8,2X))
C
C     FIND THE LARGEST PSEUDOAUTOCORRELATION
C
      XLTH2=LTH2
      XLRGE=0.
      M=1
      DO 95 I=LP1+1,LTH
      IF(XR(I).LT.XLRGE)GO TO 95
      M=I
      XLRGE=XR(I)
95    CONTINUE
C     15 AND 94 ARE THGE SMALLEST AND LARGEST ALLOWABLE
C     VALUES INTO THE PITCH QUANTIZER ROUTINES
      IF(M.LT.15)M=15
      IF(M.GT.94)M=94
      IF(PWATE.EQ.NO)IPDEC=0
      IF(IPDEC.EQ.0)M=1
C     IF M=1 THAN THE FRAME IS NOT PITCH WEIGHTED
C     THEREFORE THE 6 BITS FOR PITCH AND 2 BITS FOR
C     PITCH MAGNITUDE CAN BE ADDED TO DCT BIT ALLOCATION
      IF(M.EQ.1)BTLTH=BTLTH+8
C
C     GET PITCH GAIN
C
```

```
      G=ABS(XP(M)/XP(1))
C     COMPRESS THE PITCH GAIN
      G=SQRT(G)
D     WRITE(4,904)M,G
D904  FORMAT(/1X,'M=',I3,' AND G=',E15.8)
C
C     CREATE LPC PARAMETERS
C
      RX=R(1)
      DO 627 I=1,LP1
627   R(I)=R(I)/RX
C     <<<<<<<<<<<<<<<<<<<<<<<<<<<<
C     OVERLAY 3 COMPUTES THE PREDICTOR COEFFICIENTS OF THE WAVEFORM
C
      CALL OVR3
      CALL SOLVE(A,R,LPCN,U,PARCOR)
C     <<<<<<<<<<<<<<<<<<<<<<<<<<
D     WRITE(4,933)(I,PARCOR(I),I=1,LPCN)
D933  FORMAT(/1X,4(1X,'PARC(',I2,')=',E15.8,2X))
D     WRITE(4,906)(I,A(I),I=1,LPCN)
D906  FORMAT(/1X,5(1X,'A(',I2,')=',E15.8,2X))
      IF(U.GT.0)GO TO 640
      WRITE(5,9998)ICOUNT
9998  FORMAT(1X,'***LPC ERROR*** FRAME=',I6/)
      GO TO 5000
640   CONTINUE
      RESENG=SQRT(U*RX)
D     WRITE(4,932)RESENG
D932  FORMAT(/1X,'RESENG=',E15.8)
C
C     QUANTIZE PARCOR(J),J=1,...LPCN,DCBIAS,VAR,G,M
C
C     QUANTIZE PARCOR(J)
      DO 651 J=1,LPCN
      NL=2**BITSP(J)
      K=(J-1)*32
      DO 650 I=1,NL
C     ADD BIAS
      PARCJ=-PARCOR(J)+1.0
      K=K+1
      IF(PARCJ.LE.PTHR(K))GO TO 651
650   CONTINUE
651   QTPAR(J)=I-1
D     WRITE(4,924)(J,QTPAR(J),J=1,LPCN)
D924  FORMAT(/1X,4(1X,'QTPAR(',I2,')=',I6,2X))
C     QUANTIZE DC BIAS,DCBIAS
      NL=2**4
      DO 660 I=1,NL
      IF(DCBIAS.LE.DCTHR(I))GO TO 661
660   CONTINUE
661   QTDC=I-1
D     WRITE(4,925)QTDC
D925  FORMAT(/1X,'QTDC=',I6)
C     QUANTIZE VARIANCE,VAR
      NL=2**5
      DO 680 I=1,NL
      IF(VAR.LE.VARTHR(I))GO TO 681
680   CONTINUE
681   QTVAR=I-1
D     WRITE(4,926)QTVAR
D926  FORMAT(/1X,'QTVAR=',I6)
C     QUANTIZE PITCH GAIN,G
      NL=2**2
      DO 670 I=1,NL
670   IF(G.LE.PGTHR(I))GO TO 671
```

```
971     QTG=I-1
C       QUANTIZE PITCH,M
C       QUANTIZE THE PITCH IN 6 BITS OR 64 LEVELS
C       WITH THE HIGHEST VALUE BEING 94 AND LOWEST 15
        IBPT=0
        IF(IPDEC.EQ.0)GO TO 87
        ITR=M-15
        IBPT=ITR
        IF(ITR.LE.47)GO TO 9191
        ITR1=(ITR-46)/2
        IBPT=47+ITR1
        ITR=(ITR/2)*2
9191    CONTINUE
        DTM=ITR+15
97      QTM=IBPT
D       WRITE(4,927)QTM,QTG
D927    FORMAT(/1X,'QTM=',I6,' AND QTG=',I6)
C
C       DEQUANTIZE PARCOR(J),J=1,...LPCN,G,M
C
C       DEQUANTIZE PARCOR(J)
        DO 751 J=1,LPCN
        K=(J-1)*32+1
        I=QTPAR(J)+K
C       REMOVE BIAS
751     DTPAR(J)=-PDEC(I)+1.0
D       WRITE(4,928)(J,DTPAR(J),J=1,LPCN)
D928    FORMAT(/1X,4(1X,'DTPAR(',I2,')=',E15.8,2X))
C       DEQUANTIZE PITCH GAIN,G
        I=QTG+1
        DTG=PGDEC(I)
C       EXPAND THE PITCH GAIN
C       DTG=DTG**2
C       DEQUANTIZE PITCH,M
        IF(IPDEC.EQ.0)DTM=1
89      CONTINUE
D       WRITE(4,929)DTM,DTG
D929    FORMAT(/1X,'DTM=',E15.8,' AND DTG=',E15.8)
C
C       RECREATE LP COEFFICENTS
C       <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
C
C       CALL OVR4
        CALL PARPRE(LPCN,DTPAR,DTA)
C       <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
D       WRITE(4,931)(J,DTA(J),J=1,LPCN)
D931    FORMAT(/1X,4(1X,'DTA(',I2,')=',E15.8,2X))
C
C       REGENERATE RESIDUAL ENERGY
C
        U=1.0-DTPAR(1)**2
        DO 870 I=2,LPCN
870     U=U*(1.0-DTPAR(I)**2)
        ENG=SQRT((2.0**NSTAGE)*U)
D       WRITE(4,932)ENG
C
C       WRITE(5,6000)I
C6000   FORMAT(1X,I6)
C       FORMULATE LPC BASIS SPECTRUM
C
        XR(1)=1.
        XI(1)=0.
        DO 628 I=1,LPCN
628     XR(I+1)=DTA(I)
        DO 629 I=LPCN+2,LTH2
```

```
629     XP(I)=0.
C
C       FORMULATE THE PITCH WEIGHTING SPECTRUM
C       NOTICE WE ARE CONSTRUCTING A ZERO DC
C       PITCHED SIGNAL TO USE IN THE FREQUENCY DOMAIN
C
        DTM=DTM-1
        DO 98 I=1,LTH2
98      XI(I)=0.
        EGR=0.
C       K1=LTH/IFIX(DTM)
        DCPIT=0.
        DO 99 K=1,100
        J=(K-1)*DTM+1
        IF(J.GT.LTH)GO TO 1020
        TEMP=DTG**(K-1)
        DCPIT=DCPIT+TEMP
99      XI(J)=TEMP
1020    DCPIT=DCPIT/FLOAT(LTH)
        DO 9797 I=1,LTH
        XI(I)=XI(I)-DCPIT
        EGR=EGR+XI(I)**2
9797    CONTINUE
        EGR=1./SQRT(EGR)
D       WRITE(4,917)K,EGR
D917    FORMAT(/1X,'K=',I3,' AND EGR=',E15.8)
C       <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
C       OVERLAY 5 DETERMINS THE MAGNITUDE SPECTRUM OF THE TWO REAL SIGNALS XR,XI
C       WHICH ARE BOTH LOADED INTO ONE FFT
C       CALL POWMAG(XR,XI,NSTAGE,LTH2)
C       CALL OVRS
C
C       <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
        DO 630 I=1,LTH
630     DCT1(I)=ENG/XR(I)
D       WRITE(4,907)(I,DCT1(I),I=1,LTH)
D907    FORMAT(/1X,4(1X,'DCT1(',I3,')=',E15.8,2X))
C       PITWT IS USED TO APPROPRIATELY ADJUST
C       THE FREQUENCY DOMAIN PITCH WEIGHTING FUNCTION.
C       IT LOWPASS FILTERS IT FOR FREQUENCIES BELOW
C       1.5KHZ,SIMULATING A POLE IN THE DRIVING FUNCTION
C       IT ALSO UNCORRELATES THE FREQUENCY DOMAIN
C       COMPONENTS FOR THE HIGH FREQUENCIES SIMULATING
C       A WHITE NOISE STIMULUS IN THE HIGH FREQUENCY PART OF THE SPECTRUM.
C       IF THE FRAME IS UNPITCHED THAN JUMP AROUND IT.
        IF(DTM.EQ.0)GO TO 1009
        CALL PITWT(EGR,LTH,XI,PWEIT)
        GO TO 1011
1009    DO 1010 I=1,LTH
C       PWEIT(I)=EGR*XI(I)
1010    PWEIT(I)=1.
1011    CONTINUE
D       WRITE(4,905)(I,PWEIT(I),I=1,LTH)
D905    FORMAT(/1X,4(1X,'PWEIT(',I3,')=',E15.8,2X))
C
C       COMPLETE BASIS SPECTRUM
C
        DO 102 I=1,LTH
102     DCT1(I)=DCT1(I)*PWEIT(I)
D       WRITE(4,908)(I,DCT1(I),I=1,LTH)
D908    FORMAT(/1X,4(1X,'DCT1(',I3,')=',E15.8,2X))
        DCT1(1)=1.0E-7
C
C       ORDER DCT MAX-TO-MIN
```

```
C       <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
C       OVERLAYS 6,7 ARE JUST SORTING ALGORITHMS TO SORT THE
C       DCT COEFFICIENTS INTO THEIR DESCENDING ORDER
        IF(TYPSRT.EQ.YES)CALL OVR6
        IF(TYPSRT.EQ.NO)CALL OVR7
C       <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
D       WRITE(4,909)(I,DCT2(I),I=1,LTH)
D909    FORMAT(/1X,4(1X,'DCT2(',I3,')=',E15.8,2X))
D       WRITE(4,910)(I,IORDR(I),I=1,LTH)
D910    FORMAT(/1X,6(1X,'IORDR(',I3,')=',I3,2X))
C
C       COMPRESS BASIS SPECTRUM
C
        DO 4 I=1,LTH
        IF(DCT2(I).EQ.0)DCT2(I)=10E-10
4       DCT2(I)=ALOG(DCT2(I))/DLOG2
D       WRITE(4,909)(I,DCT2(I),I=1,LTH)
C
C       INITIAL DCT BIT ASSIGNMENT
C
        TYPE 603,BTLTH
603     FORMAT(1X,'NUMBER OF BITS FOR DCT=',1X,F11.4)
        SLOG=0.
        DO 104 I=1,LTH
104     SLOG=SLOG+DCT2(I)
C       CLEAR OUT BIT-ASSIGNMENT ARRAY
        DO 105 I=1,LTH
105     IBIT(I)=0
        CTEMP=(BTLTH-SLOG)/LTH
        DO 1050 I=1,LTH
        BITS=CTEMP+DCT2(I)
        IF(BITS.LE.0.0)GO TO 1051
1050    LAST=I
C       LAST POSITIVE ENTRY IS LAST SAMPLE TO BE USED
1051    ITOT=LAST
D       WRITE(4,930)BITS,CTEMP,ITOT
D930    FORMAT(/1X,'BITS=',E15.8,' W/ CTEMP=',E15.8,' @ ITOT=',I3)
        SLOG=0.0
C       FORM SUM-OF-LOGS UP TO LAST SAMPLE
        DO 109 I=1,ITOT
109     SLOG=SLOG+DCT2(I)
        CTEMP=(BTLTH-SLOG)/ITOT
C       TOTAL ASSIGNABLE BITS
        IBITSL=BTLTH
        DO 110 I=1,ITOT
        K=I
        BITS=CTEMP+DCT2(I)
C       ROUND UPWARDS
        IF(BITS.LT.0.0)BITS=0
        IBITS=BITS+0.5
C       SET BIT-ASSIGNMENT & CLAMP @ 5 BITS
        IF(IBITS.GE.5)IBITS=5
        IBIT(I)=IBITS
C       REMAINING BITS=LAST VALUE-NEWEST ASSIGNMENT
        IBITSL=IBITSL-IBITS
        IF(IBITSL.GT.0)GO TO 110
C       EITHER NONE REMAINING OR TOO MANY IN LAST ASSIGNMENT
        IBIT(I)=IBIT(I)+IBITSL
        GO TO 111
110     CONTINUE
C
C       FINAL DCT BIT ASSIGNMENT
C
C       ALL LTH SAMPLES USED.TRY TO SCALE UPWARDS
        DO 160 I=1,LTH
```

```
      IF(IBITSL.LE.0)GO TO 111
      IBITI=IBIT(I)+1
C     CLAMP @ 5 BITS/SAMPLE
      IF(IBITI.GT.5)GO TO 160
      IBIT(I)=IBITI
      IBITSL=IBITSL-1
160   CONTINUE
111   CONTINUE
D     WRITE(4,911)(I,IBIT(I),I=1,LTH)
D911  FORMAT(/1X,6(1X,'IBIT(',I3,')=',I3,2X))
C
C     QUANTIZE DCT(I)
C
      DO 113 I=1,LTH
      J=IORDR(I)
      IQTDCT=0
      NUMBIT=IBIT(I)
      LEVEL=2**NUMBIT
      IF(NUMBIT.LE.0)LEVEL=0
      NL2=LEVEL/2
      K=NUMBIT*16
      DO 115 L=1,NL2
      K=K+1
      IF(ABS(DCT(J)).LE.DCTTHR(K)*DCT1(J))GO TO 114
115   CONTINUE
114   IQTDCT=L-1
      IF(DCT(J).LT.0.0)IQTDCT=IQTDCT+NL2
C     QTDCT(J)=IQTDCT
113   QTDCT(I)=IQTDCT
D     WRITE(4,912)(I,QTDCT(I),I=1,LTH)
D912  FORMAT(/1X,4(1X,'QTDCT(',I3,')=',I3,2X))
      IF(SER.EQ.NO)GO TO 1902
C
C     <<<<<<<<<<<<<<<<<<<<<<<<<
C     >>>>>>>  <<<<<<<<
C     >>>>>>>>>>>>>>>>>>>>>>>>>
C
C     NOW BEGIN DECIMAL TO DIGITAL CONVERSION AND ERRROR
C     PROTECTION TO ALLOW FOR MODEM TRANSMISSION
C
      CALL OVR8(NBPF,NBRPF)
C     SUBROUTINE DITBA(NBPF)
C     THESE BITS IN THE INBA VECTOR WILL BE
C     TRANSMITTED THROUGH A CHANNEL WHICH IS NOT ERROR FREE
C
      IF(ENCDEC.EQ.NO)GO TO 4449
C     <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
C     <<<<<<<<<<<<<<<CHANNEL>>>>>>>>>>>>>
C     <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
C     FIRST ENCODE IN AN MBCH(63,45) CODE
C     THEN ENCODE 3 BLOCKS WORTH OF DATA
C     THE TOTAL NUMBER OF BITS ERROR PROTECTED IS 3*45
      DO 446 NDB=1,NEPB
      CALL OVR9
C     CALL ENCBCH
446   CONTINUE
4449  IF(ERRDEC.EQ.NO)GO TO 4450
C
C
C     <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
C     SIMULATE CHANNEL WITH DESIRED ERROR RATE OF PREA
C
      WRITE(5,1905)(INBA(I),I=1,NBRPF)
      CALL OVR13(NBRPF,PREA,IRN,JRN,NERB,3)
C     CALL CEIR(    )
```

```
4450    IF(ENCDEC.EQ.NO)GO TO 449
C
C       <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
C       NOW DECODE AND DETECT ERROR LOCATION
C
C       WRITE(5,1905)(INBA(I),I=1,NBRPF)
1905    FORMAT(1X,63(I1))
        NFRSK=0
        DO 448 NDB=1,NEPB
C       CALL DECBCH
        CALL OVR10
        IF(NDB.EQ.1.AND.NES.GE.4)NFRSK=1
448     CONTINUE
C       WRITE(5,1905)(INBA(I),I=1,NBRPF)
449     CONTINUE
C       <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
C       <<<<<<<<<<<<<<RECEIVER<<<<<<<<<<<<<<<<<<<
C       <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
C
C       NOW CREATE DECIMAL DATA FROM THE BINARY DATA
C
C       READ SIDE INFORMATION FROM BINARY VECTOR
C       <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
C
        CALL OVR11
C       CALL BATSD(NEPB)
C       <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
1902    CONTINUE
        DO 1900 J=1,LPCN
1900    QRPAR(J)=QTPAR(J)
        QRG=QTG
        QRM=QTM
        QRDC=QTDC
        QRVAR=QTVAR
C       DO 1901 I=1,LTH
C1901   QRDCT(I)=QTDCT(I)
C
C
C       DEQUANTIZE PARCOR(J),J=1,...LPCN,G,M,DCBIAS,VAR
C
C       DEQUANTIZE PARCOR(J)
        DO 1751 J=1,LPCN
        K=(J-1)*32+1
        I=QRPAR(J)+K
C       REMOVE BIAS
1751    DRPAR(J)=-PDEC(I)+1.0
D       WRITE(4,940)(J,DRPAR(J),J=1,LPCN)
D940    FORMAT(/1X,4(1X,'DRPAR(',I2,')=',E15.8,2X))
C       DEQUANTIZE PITCH GAIN.G
        I=QRG+1
        DRG=PGDEC(I)
C       DEQUANTIZE PITCH,M
        IF(IPDEC.EQ.0)GO TO 9193
        ITR=QRM
        IF(ITR.LE.47)GO TO 9192
        ITR1=QRM-47
        ITR=(ITR1-1)*2+48
9192    DRM=ITR+15
        GO TO 9194
9193    DRM=1
9194    CONTINUE
D       WRITE(4,941)DRM,DRG
D941    FORMAT(/1X,'DRM=',E15.8,' AND DRG=',E15.8)
C       DEQUANTIZE DC BIAS,DCBIAS
        I=QRDC+1
```

```
      DRDC=DCDEC(I)
D     WRITE(4,942)DRDC,QRDC
D942  FORMAT(/1X,'DRDC=',E15.8,5X,'QRDC=',I5)
C     DEQUANTIZE VARIANCE,VAR
      I=QRVAR+1
      DRVAR=VARDEC(I)
D     WRITE(4,943)DRVAR
D943  FORMAT(/1X,'DRVAR=',E15.8)
C
C     RECREATE LP COEFFICENTS
C
C     <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
      CALL OVR4
      CALL PARPRE(LPCN,DRPAR,DRA)
C
C     <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
C     REGENERATE RESIDUAL ENERGY
C
      U=1.0-DRPAR(1)**2
      DO 1870 I=2,LPCN
1870  U=U*(1.0-DRPAR(I)**2)
      ENG=SQRT((2.0**NSTAGE)*U)
D     WRITE(4,932)ENG
C
C     FORMULATE LPC BASIS SPECTRUM
C
      XR(1)=1.
C     XI(1)=0.
      DO 1628 I=1,LPCN
1628  XR(I+1)=DRA(I)
C1628 XI(I+1)=0.
      DO 1629 I=LPCN+2,LTH2
1629  XR(I)=0.
C
C     FORMULATE THE PITCH WEIGHTING SPECTRUM
C     AGAIN WE WILL CONSTRUCT A ZERO DC
C     BIASED PITCHED SIGNAL
C
      DRM=DRM-1
      DO 198 I=1,LTH2
C     XR(I)=0.
198   XI(I)=0.
C     K1=LTH/IFIX(DRM)
      DCPIT=0.0
      EGR=0.
      DO 199 K=1,100
      J=(K-1)*DRM+1
      IF(J.GT.LTH)GO TO 11020
      TEMP=DRG**(K-1)
      DCPIT=DCPIT+TEMP
199   XI(J)=TEMP
11020 DCPIT=DCPIT/FLOAT(LTH)
      DO 9798 I=1,LTH
      XI(I)=XI(I)-DCPIT
      EGR=EGR+XI(I)**2
9798  CONTINUE
      EGR=1./SQRT(EGR)
D     WRITE(4,917)K,EGR
C     <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
      CALL OVR5
C     CALL POWMAG(XR,XI,NSTAGE,LTH2)
C     <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
      DO 1630 I=1,LTH
1630  DCT1(I)=ENG/XR(I)
D     WRITE(4,907)(I,DCT1(I),I=1,LTH)
```

```
      IF(DRM.EQ.0)GO TO 11009
      CALL PITWT(EGR,LTH,XI,PWEIT)
      GO TO 11011
11009 DO 11010 I=1,LTH
C     PWEIT(I)=EGR*SQRT(XR(I)*XR(I)+XI(I)*XI(I))
C     PWEIT(I)=FGR*XI(I)
11010 PWEIT(I)=1.0
11011 CONTINUE
D     WRITE(4,905)(I,PWEIT(I),I=1,LTH)
C
C     COMPLETE BASIS SPECTRUM
C
      DO 1102 I=1,LTH
1102  DCT1(I)=DCT1(I)*PWEIT(I)
D     WRITE(4,908)(I,DCT1(I),I=1,LTH)
C     ORDER DCT MAX-TO-MIN
C     <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
C
      DCT1(1)=1.E-7
      IF(TYPSRT.EQ.YES)CALL OVR6
      IF(TYPSRT.EQ.NO)CALL OVR7
C     <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
D     WRITE(4,909)(I,DCT2(I),I=1,LTH)
D     WRITE(4,910)(I,IORDR(I),I=1,LTH)
C
C     COMPRESS BASIS SPECTRUM
C
      DO 14 I=1,LTH
      IF(DCT2(I).EQ.0)DCT2(I)=10E-10
14    DCT2(I)=ALOG(DCT2(I))/DLOG2
D     WRITE(4,909)(I,DCT2(I),I=1,LTH)
C
C     INITIAL DCT BIT ASSIGNMENT
C
      SLOG=0.
      DO 1104 I=1,LTH
1104  SLOG=SLOG+DCT2(I)
C     CLEAR OUT BIT-ASSIGNMENT ARRAY
      DO 1105 I=1,LTH
1105  IBIT(I)=0
      CTEMP=(BTLTH-SLOG)/LTH
      DO 11050 I=1,LTH
      BITS=CTEMP+DCT2(I)
      IF(BITS.LE.0.0)GO TO 11051
11050 LAST=I
C     LAST POSITIVE ENTRY IS LAST SAMPLE TO BE USED
11051 ITOT=LAST
D     WRITE(4,930)BITS,CTEMP,ITOT
      SLOG=0.0
C     FORM SUM-OF-LOGS UP TO LAST SAMPLE
      DO 1109 I=1,ITOT
1109  SLOG=SLOG+DCT2(I)
      CTEMP=(BTLTH-SLOG)/ITOT
C     TOTAL ASSIGNABLE BITS
      IBITSL=BTLTH
      DO 1110 I=1,ITOT
      K=I
      BITS=CTEMP+DCT2(I)
C     ROUND UPWARDS
      IF(BITS.LT.0.0)BITS=0
      IBITS=BITS+0.5
C     SET BIT-ASSIGNMENT & CLAMP @ 5 BITS
      IF(IBITS.GE.5)IBITS=5
      IBIT(I)=IBITS
C     REMAINING BITS-LAST VALUE-NEWEST ASSIGNMENT
```

```
      IBITSL=IBITSL-IBITS
      IF(IBITSL.GT.0)GO TO 1110
C     EITHER NONE REMAINING OR TOO MANY IN LAST ASSIGNMENT
      IBIT(I)=IBIT(I)+IBITSL
      GO TO 1111
C     BITS STILL REMAIN TO BE ASSIGNED
1110  CONTINUE
C
C     FINAL DCT BIT ASSIGNMENT
C
C     ALL LTH SAMPLES USED,TRY TO SCALE UPWARDS
      DO 1160 I=1,LTH
      IF(IBITSL.LE.0)GO TO 1111
      IBITI=IBIT(I)+1
C     CLAMP @ 5 BITS/SAMPLE
      IF(IBITI.GT.5)GO TO 1160
      IBIT(I)=IBITI
      IBITSL=IBITSL-1
1160  CONTINUE
1111  CONTINUE
D     WRITE(4,911)(I,IBIT(I),I=1,LTH)
      IF(SER.EQ.NO)GO TO 1115
C
C
C     <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
C     CALL THE BINARY TO DECIMAL CONVERSION ROUTINE FOR
C     DCT COEFFICIENTS
C
      CALL OVR12(NBPF)
C     CALL SBADCT(NBPF)
C     <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
      DO 1112 I=1,LTH
1112  QTDCT(I)=IFIX(DCT2(I))
1115  DO 1116 I=1,LTH
1116  QRDCT(I)=QTDCT(I)
D     WRITE(4,935)(I,QRDCT(I),I=1,LTH)
D935  FORMAT(/1X,4(1X,'QRDCT(',I3,')=',I3,2X))
C
C     DEQUANTIZE DCT(I)
C
      DO 1113 I=1,LTH
      J=IORDR(I)
      RDCTI=0.0
      NUMBIT=IBIT(I)
      LEVEL=2**NUMBIT
      NL2=LEVEL/2
      K=NUMBIT*16
      SIGN=1.0
C     INDEX=QRDCT(J)
      INDEX=QRDCT(I)
      IF(INDEX.LT.NL2)GO TO 1114
      SIGN=-1.0
      INDEX=INDEX-NL2
1114  L=INDEX+K+1
      RDCTI=DCTDEC(L)
      DRDCT(J)=SIGN*RDCTI*DCT1(J)
1113  CONTINUE
D     WRITE(4,934)(I,DRDCT(I),I=1,LTH)
D934  FORMAT(/1X,4(1X,'DRDCT(',I3,')=',E15.8,2X))
C
C     WE WILL USE THE PREVIOUS FRAME SIGNAL IF BURST
C     ERRORS WERE DETECTED INTHE FIRST ERROR
C     PROTECTED BLOCK
      IF(NFRSK.EQ.1)GO TO 1061
C>>>>>>>>>>>>>>>>>>>>>>
```

```
C>>>>>>>>>>
C
C       INVERSE COSINE TRANSFORM
C
C
C       NOW LOAD DRDCT IN.O DCT BUFFER FOR INVERSE TRANSFORM
C
        DO 125 I=1,LTH
125     DCT(I)=DRDCT(I)
        INIT1=2
C       <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
C       NOW DO THEINVERSE DCT AS INIT1=2
        CALL OVR1
        CALL DCTSUB(LTH,XR,XI,DRDCT,2)
C       <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
        DO 126 I=1,LTH
        Y(LTH+I)=XR(LTH+I)
126     Y(I)=XR(I)
D       WRITE(4,914)(I,Y(I),I=1,LTH)
D914    FORMAT(/1X,4(1X,'Y(',I3,')=',E15.8,2X))
C
C       EXPAND STATISTICS
C
        DRVAR=10.0**(DRVAR/20.0)
        DRDC=DRDC*DRVAR
D       WRITE(4,942)DRDC
D       WRITE(4,943)DRVAR
C
C       RENORMALIZE BY STATISTICS
C
        DO 1120 J=1,LTH
        Y(J)=DRVAR*Y(J)
1120    Y(J)=Y(J)+DRDC
D       WRITE(4,914)(I,Y(I),I=1,LTH)
C
C       INTERPOLATE OVER NP POINTS
C
        DO 1122 I=1,NP
1122    Y(I)=YY(I)*(XN-I+1)/XN+Y(I)*(I-1)/XN
C
C       NOW USE PREVIOUS FRAME FOR 4 ERRORS IN BLOCK ONE
C
        GO TO 1062
1061    CONTINUE
C       FIRST REVERSE SIGNAL TO MAKE IT SYMMETRIC AND
C       REDUCE THE PHASE DISCONTINUITY AT THE BOUNDARY
        LTHH=LTH/2
        DO 1063 I=1,LTHH
        IXX=Y(I)
        Y(I)=Y(LTH+1-I)
        Y(LTH+1-I)=IXX
1063    CONTINUE
1062    CONTINUE
C       SHIFT NECESSARY INFO FOR NEXT FRAME
C
        DO 389 I=1,NP
389     YY(I)=Y(LTH+I-NP)
C
C       DATA OUTPUT
C
        DO 123 I=1,NTOTO
123     NOUT(I)=Y(I)
        CALL TAPE3(2)
D       WRITE(4,913)(I,NOUT(I),I=1,LTH)
```

```
0913   FORMAT(/1X,4(1X,'NOUT(',I3,')=',I6,2X))
C      ICSUB IS USED IN THE SIGNAL TO NOISE
C      CALCULATION. IT CORRESPONDS TO THE NUMBER
C      OF FRAME WHICH HAD MOPE THAN 3 ERRORS
C      IN THE FIRST BLOCK.
       IF(NFRSK.EQ.0)GO TO 1065
       ICSUB=ICSUB+1
       GO TO 1066
C
C      <<<<<<<<<<<<<<<<<<<<<<<<<<
C      >>>>> END OF PROCESS <<<<<
C      >>>>>>>>>>>>>>>>>>>>>>>>>>
C
C      COMPUTE S/N RATIO
C
1065   CONTINUE
       SUM1=0.
       SUM2=0.
       DO 3605 I=1,NTOTO
       XNIN=FLOAT(NIN(I))
       SUM1=SUM1+XNIN*XNIN
3605   SUM2=SUM2+(XNIN-FLOAT(NOUT(I)))**2
       SNR=10.*ALOG10(SUM1/SUM2)
       FRAME=ICOUNT-ICSUB-INITFR+1-IVARF
       IBTLTH=INT(BTLTH)
       CSN=((FRAME-1.)/FRAME)*CSN+(1./FRAME)*SNR
       WRITE(5,255)ICOUNT,SNR,CSN,M,G,IBTLTH
255    FORMAT(1X,'FRAME=',I4,2X,'SN=',F5.2,2X,'CSN=',F5.2,2X,'PITCH=',I3
      1,2X,'P.GAIN=',F6.3,2X,'DCT BITS=',I4/)
1066   CONTINUE
       GO TO 1001
5000   CONTINUE
9999   STOP 'ATC70 DONE !'
       END
       SUBROUTINE PITWT(EGR,LTH,XI,PWEIT)
       DIMENSION XI(1),PWEIT(1)
       STRR=0.0
       DO 1010 I=1,LTH
       T1=EGR*XI(I)
       T2=FLOAT(I-1)/FLOAT(LTH)
       T3=1.0-T2
       XI(I)=T1*T3+T2
       IF(XI(I).LE.0.5)XI(I)=0.5
       STRR=STRR+XI(I)**2
1010   CONTINUE
       STRR=SQRT(FLOAT(LTH)/STRR)
       DO 1011 I=1,LTH
1011   PWEIT(I)=XI(I)*STRR
       RETURN
       END
      SUBROUTINE OVP1
C     SUBROUTINE DCTSUB(LTH,YR,YI,DCT,INIT1)
C     THIS PROGRAMS DOES THE FORWARD AND INVERSE FAST
C     DISCRETE COSINE TRANSFORMS. IF THE VALUE OF INIT1
C     IS 1 THEN THE FORWARD DCT IS DONE.  IN THIS CASE
C     LTH DCT COEFFICIENTS ARE RETURNED ALONG WITH 2LTH
C     MAGNITUDE SPECTRUM COEFFICIENTS TO BE USED
C     IN THE AUTOCORRELATION FUNCTION ESTIMATION.
C
C     IF THE VALUE OF INIT1 IS 2 THEN THE INVERSE
C     DCT IS PERFORMED AND LTH VALUES OF A DISCRETE TIME SIGNAL ARE
C     RETURNED.
      COMMON/BLOCK1/INIT1,DCT(256),LTH,NSTAGE
      COMMON/SORT/DCT1(256),DCT2(256),IORDR(256),YR(512),YI(512)
      LTH1=LTH/2
      LTH2=LTH1+2
      LTH3=LTH+2
      L2TH=2*LTH
      LTHALF=LTH1/2
      PI=4.0*ATAN(1.0)
      PI2=2*PI/LTH
      PI3=PI2/8
      PI6=2*PI3
      IF(INIT1.EQ.2) GO TO 4000
C     NOW RESHUFFLE BY FIRST ADDING LTH ZEROES
C     AND THEN RESHUFFLING TO DO THE FFT
      DO 10 I=1,LTH
      YI(I+LTH)=0.0
10    YR(LTH+I)=0.0
      DO 11 I=1,LTH
      YI(I)=YR(2*I-1)
11    YI(2*LTH+1-I)=YR(2*I)
      DO 12 I=1,2*LTH
      YR(I)=YI(I)
12    YI(I)=0.0
C
C     NOW SEPARATE INTO EVEN AND ODD COMPONENTS AND SCALE BY 2
C
      DO 13 J=1,LTH
      YR(J)=YR(2*J-1)/2.0
13    YI(J)=YR(2*J)/2.0
      CALL OVEVOD(LTH,1,1)
C     CALL EVODD(LTH,YR,YI,1,1)
      DO 35 I=1,LTH
      DCT(I)=2*(COS((2*I-2)*PI3)*YR(2*I-1)+SIN((2*I-2)*PI3)*YI(2*I-1))
      DCTEMP=2*(COS((2*I-1)*PI3)*YR(2*I)+SIN((2*I-1)*PI3)*YI(2*I))
      YR(2*I-1)=DCT(I)**2
      YR(2*I)=DCTEMP**2
      YI(2*I-1)=0.0
35    YI(2*I)=0.0
      GO TO 5000
C
C     NOW WE WILL DO THE INVERSE DCT
4000  CONTINUE
C     TYPE506,INIT1,LTH,NSTAGE
C506  FORMAT(1X,I4,1X,I4,1X,I4)
C     WRITE(5,934)(I,DCT(I),I=1,LTH)
C934  FORMAT(/1X,4(1X,'DRDCT2(',I3,')=',E15.8,2X))
      YR(1)=DCT(1)/2.0
      YI(1)=0.0
      YR(LTH/2+1)=(COS(PI6*LTH/2)*DCT(LTH/2+1)+SIN(PI6*LTH/2)*
     1DCT(LTH/2+1))/2.0
      YI(LTH/2+1)=(SIN(PI6*LTH/2)*DCT(LTH/2+1)-COS(PI6*LTH/2)
     1*DCT(LTH/2+1))/2.0
```

```
C
      DO 410 K=2,LTH/2
      YR(K)=(COS(PI6*(K-1))*DCT(K)+SIN(PI6*(K-1))*DCT(LTH+2-K))/2.0
      YI(K)=(SIN(PI6*(K-1))*DCT(K)-COS(PI6*(K-1))*DCT(LTH+2-K))/2.0
      YR(LTH+2-K)=YR(K)
410   YI(LTH+2-K)=-1.0*YI(K)
C
      DO 420 K=1,LTH/4+1
      J=K-1
      Q1=(YI(K)+YI(LTH/2+2-K))/2.
      Q2=(YI(K)-YI(LTH/2+2-K))/2.
      Q3=(YR(K)+YR(LTH/2+2-K))/2.
      Q4=(YR(K)-YR(LTH/2+2-K))/2.
C
      QTEMP1=SIN(2*PI*J/LTH)*Q4
      QTEMP2=SIN(2*PI*J/LTH)*Q1
      QTEMP3=COS(2*PI*J/LTH)*Q1
      QTEMP4=COS(2*PI*J/LTH)*Q4
C
      YR(K)=Q3-QTEMP1-QTEMP3
      YI(K)=Q2+QTEMP4-QTEMP2
      YR(LTH/2+2-K)=Q3+QTEMP1+QTEMP3
426   YI(LTH/2+2-K)=QTEMP4-QTEMP2-Q2
C     WRITE(S,934)(I,YR(I),I=1,LTH)
C     WRITE(S,934)(I,YI(I),I=1,LTH)
      CALL FASTF(NSTAGE-2,2,1,1)
      DO 430 N=1,LTH/2
      DCT(2*N-1)=YR(N)
      DCT(2*N)=YI(N)
      YI(N)=0.0
430   YI(LTH/2+N)=0.0
C
      DO 445 I=1,LTH/2
      YR(2*I-1)=DCT(I)
445   YR(2*I)=DCT(LTH-I+1)
5000  RETURN
      END
```

```
C       OVR2.FTN        27-NOV-79
C
        SUBROUTINE OVR2
        COMMON/MTAPE0/NIN(256),NOUT(256)
        COMMON/MTAPE1/NSKIP,IST,NTOTI,NTUPS,NTOTO
        COMMON/MTAPE2/NEND,NERR,NFILE,NINS,NOUTS
        COMMON/MTAPE3/NBF(1324),NBUF(1324)
        COMMON/MTAPE4/LST,IBEG
        COMMON/MTAPE5/MASK,ISW(2),IOATT,IOSUC,IEALN,IORWD
        1,IOWLB,IVER,IOSPF,IEEOF,IOZOF,IORLB,MT0(6),MT1(6),DSW
        COMMON/MTAPE6/APPEND
        COMMON/SORT/DCT1(256),DCT2(256),IORDR(256),XR(512),XI(512)
C       NOW INSERT COMMON BLOCKS FOR THE OVERLAYS
        COMMON/BLOCK1/INIT1,DCT(256),LTH,NSTAGE
        COMMON/BLOCK2/A(8),R(11),U,PARCOR(8)
        COMMON/BLOCK3/DTPAR(8),DTA(8),LPCN
        EQUIVALENCE  (DCT,DRDCT),(DTPAR,DRPAR),(DTA,DRA)
        COMMON/BLOCK4/LTH2
        COMMON/BLOCK5/LTH3,LTH4,NP,XN,LP1,ARG
        1,DLOG2,CNS,ICOUNT,IREC,NBPF,NBPPF
        2,PI,NSTAG,BTLTH,PWATE,TYPSRT,NEPB
        LOGICAL*1 PWATE,YES,NO,TYPSRT
        DATA YES/'Y'/
        DATA NO/'N'/
C
C       INITIALIZATION
C
C       WRITE(5,2342)
C2342   FORMAT(1H$,'NUMBER OF BITS/FRAME TOTAL=')
C       READ(5,101,END=9999,ERR=9999)NBPF
C       WRITE(5,2345)
C2345   FORMAT(1H$,'NUMBER OF BITS/FRAME FOR DCT=')
C       READ (5,101,END=9999,ERR=9999)NBITS
C       BTLTH=NBITS
        WRITE(5,2606)
2606    FORMAT(1H$,'USE PITCH WEIGHTING(Y/N)? ')
        READ(5,2601,END=9999,ERR=9999)PWATE
2601    FORMAT(A1)
        TYPE 2607
2607    FORMAT(1H$,'USE FAST SORT(Y/N)? ')
        READ(5,2601,END=9999,ERR=9999)TYPSRT
C
C       <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
C       <<<<<<<<<<<<<<<<<<<<<CONSTANTS
C       <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
C       THE FRAME LENGTH IS LTH AND IS 256
C       LTH=256
C       WE WILL USE PITCH WEIGHTING PWATE
C       PWATE=YES
C       WE WILL USE FASTF LINKED LIST SORT TYPSRT
C       TYPSRT=YES
C       WE WILL SET THE NUMBER OF STAGES FOR THE DFT TO
C       BE LOG(256)+1
C       NSTAGE=9
C       SET UP CONSTANTS
C
C       # OF FFT STAGES,STAGE
C       WRITE(5,2343)
C2343   FORMAT(1H$,'NUMBER OF FFT STAGES=')
C       READ(5,101,END=9999,ERR=9999)NSTAGE
        NSTAG=NSTAGE+1
C       VERY ACCURATE VALUE FOR PI=3.14159...
        PI=4.0*ATAN(1.0)
C       FRAME SIZE,LTH
```

```
C       WRITE(5,2344)
C2344   FORMAT(1H$,'FRAME SIZE IN SAMPLES=')
C       READ(5,101,END=9599,ERR=9900)LTH
        LTH2=2*LTH
        LTH3=3*LTH
        LTH4=4*LTH
C       INPUT RECORD SIZE,NTOTI
        NTOTI=LTH
C       INPUT UPDATE SIZE,NTUPS
        NTUPS=LTH-10
C       OUTPUT RECORD SIZE,NTOTO
        NTOTO=NTUPS
C       OUTPUT INTEROLATING SIZE,NP
        NP=10
        XN=NP
C       LPC ORDER
C       LPCN=8
C       LP1=LPCN+1
C       ARGUMENT FOR TRIG FUNCTIONS
        ARG=PI*(2.*LTH-1.)/(2.*LTH)
C       FOR BIT ALLOCATION STRATEGY
        DLOG2=ALOG(2.0)
C       INIT CUMULATIVE SNR RATIO
        CNS=0.
C       FRAME COUNTER
        ICOUNT=0
C
C       <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
C       <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
C       NOW SET UP BIT RATE AND BIT DEDICATION
C       FOR DCT AND SIDE INFORMATION
C       NBRPF WIL BE TOTAL NUMBER OF BITS PER FRAME
C       INCLUDING PARITY BITS
C       NTBR WIL BE BIT RATE PER SECOND
C       NSAMP WIL BE SAMPLING RATE
C       NBPF WILL BE BITS MINUS PARITY BITS
C       BTLTH=NBITS ARE BITS DEDICATED TO DCT COEFFICIENTS
C
        WRITE(5,6700)
6700    FORMAT(1H$,'SAMPLING RATE IN IG=')
        READ(5,6300)NSAMP
6300    FORMAT(I6)
C
        NTBR=9600
        BRPF=(FLOAT(NTBR)*FLOAT(NTUPS))/FLOAT(NSAMP)
        NBRPF=INT(BRPF)
C
C       NEPB IS THE NUMBER OF ERROR PROTECTED(63,45)BLOCKS
        NEPB=3
C       NBPF=NBRPF-NEPB*18 AS THERE ARE 18 PARITY BITS PER BLOCK PROTECTED
        NBPF=NBRPF-NEPB*18
C       47 BITS BITS GO TO SIDE INFORMATION WHEN FRAME IS VOICED
        NBITS=NBPF-47
C       NOTE IF FRAMEIS UNVOICED THE DCT NBITS WILL BE ADJUSTED LATER
        BTLTH=NBITS
C
C       <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
C       <<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<<
C       GET SPEECH INPUT FILE NAMES AND SET UP HANDLER
C
        CALL TAPE3(8)
        NSKIPS=NSKIP
101     FORMAT(I6)
        WRITE(5,2005)
2005    FORMAT(1H$,'NO FRAMES= ')
```

```
      READ (5,101,END=9999,ERR=9999)IREC
      RETURN
9999  CALL EXIT
      END
```

```
C       PROGRAM SOLVF.FTN
C       COMPUTES THE PREDICTOR COEFFICIENTS OF A WAVEFORM
C       GIVEN THE NORMALIZED AUTOCORRELATION COEFFICIENTS
C
        SUBROUTINE OVR3
        COMMON/BLOCK2/A(8),R(11),U,PARCOR(8)
        COMMON/BLOCK3/DTPAR(8),DTA(8),N
        DIMENSION   B(40)
C       SUBROUTINE SOLVE(A,R,N,U,PARCOR)
        A(1)=-R(2)/R(1)
        PARCOR(1)=-A(1)
        U=1.+A(1)*R(2)
D       TYPE 999,1,U
D999    FORMAT(1X,'U(',I2,')= ',E15.8)
        DO 30 I=2,N
        W=R(I+1)
        DO 10 M=1,I-1
        B(M)=A(I-M)
10      W=W+B(M)*R(M+1)
        AK=-W/U
        DO 20 M=1,I-1
20      A(M)=A(M)+AK*B(M)
        A(I)=AK
        PARCOR(I)=-AK
        U=U+AK*W
D       TYPE 999,I,U
30      CONTINUE
D       TYPE 998,(I,PARCOR(I),I=1,N)
D998    FORMAT(1X,'PARCOR(',I2,')= ',E15.8)
D       TYPE 997,(I,A(I),I=1,N)
D997    FORMAT(1X,'A(',I2,')= ',E15.8)
        RETURN
        END
        SUBROUTINE OVR4
C       SUBROUTINE PARPRE(N,PARCOR,A)
        COMMUN/BLOCK3/PARCOR(8),A(8),N
        DIMENSION  AP(50)
        A(1)=-PARCOR(1)
        DO 120 I=2,N
        IM1=I-1
        DO 110 J=1,IM1
110     AP(J)=A(J)-PARCOR(I)*A(I-J)
        AP(I)=-PARCOR(I)
        DO 140 J=1,I
140     A(J)=AP(J)
120     CONTINUE
D       TYPE 997,(I,A(I),I=1,N)
D997    FORMAT(1X,'A(',I2,')= ',E15.8)
        RETURN
        END
```

```
      SUBROUTINE OVRS
      SUBROUTINE FOWMAG(XR,XI,NSTAGE,LTH2)
C     PROGRAM TO DETERMINE POWER SPECTRUM OF TWO REAL SIGNALS
C     WE WILL USE THE PROPERTIES OF EVEN ODD
C     SEPARATION TO OBTAIN THE 2 POWER SPECTRUMS
C     X(N)=XR(N)+JXI(N)
      COMMON/SORT/DCT1(256),DCT2(256),IORDR(256),XR(512),XI(512)
      COMMON/BLOCK1/INIT,DCT(256),LTH,NSTAGE
      COMMON/BLOCK4/LTH2
      LTH3=LTH2+2
      IP=LTH2/2
      CALL FASTF(NSTAGE,1,1,2)
      XR(1)=ABS(XR(1))
      XI(1)=ABS(XI(1))
      XR(IP+1)=ABS(XR(IP+1))
      XI(IP+1)=ABS(XI(IP+1))
      DO 30 I=2,IP
      QRTEMP=2*XR(I)*XR(LTH3-I)
      QITEMP=2*XI(I)*XI(LTH3-I)
      XRTEMP=SQRT(XR(I)**2+XR(LTH3-I)**2+QRTEMP+XI(I)**2+XI(LTH3-I)
     1**2-QITEMP)
      XITEMP=SQRT(XI(I)**2+XI(LTH3-I)**2+QITEMP+XR(LTH3-I)**2+XR(I)
     1**2-QRTEMP)
      XR(I)=XRTEMP/2.0
      XI(I)=XITEMP/2.0
      XR(LTH3-I)=XR(I)
      XI(LTH3-I)=XI(I)
30    CONTINUE
      RETURN
      END
```

```
C     SORSLAV SORTS 128 DCT COEFFICIENTS INTO THEIR
C     DECREASING ORDER AND PASSES BACK A 16 BIT WORD
C     TO THE MAIN PROGRAM ATC70.
C     WRITTEN JULY 3,1979 BY MIH

      SUBROUTINE OVR6
C     SUBROUTINE  SORT4
      DIMENSION  QUANT(4),P(4)
      COMMON/SORT/DCT1(256),DCT2(256),IORDR(256),XR(512),XI(512)
      COMMON/BLOCK1/INIT1,DCT(256),LTH,NSTAGE
      INTEGER P
      REAL  YMIMR(1024)
      EQUIVALENCE  (YMIMR(1),XR(1)),(YMIMR(513),XI(1))
C     THESE QUANTAL LEVELS ARE 4 EQUAL MASS POINTS FROM THE GAMMA
C     DISTRIBUTED DCT1 COEFFICIENTS. THEY EACH REPRESENTS 25 PERCENT
C     OF THE DISTRIBUTION.
      DATA   QUANT/0.713,2.50,7.50,1.0E20/
      DO 1  L=1,4
1     P(L)=0
      DO 2 I=1,512
2     YMIMR(I)=0
      DO 5 J=1,LTH
      K=J-1
      DO 3 LEVEL=1,4
      IF(DCT1(J).LT.QUANT(LEVEL)) GO TO 4
3     CONTINUE
4     YMIMR(J+(LEVEL-1)*LTH)=FLOAT((J-1-P(LEVEL)))*256.+FLOAT(K)
5     P(LEVEL)=J
      IND=LTH
      DO 6 LEVEL=1,4
      ISKIP=0
      MARK=P(LEVEL)
7     MARK=MARK-ISKIP
      IF(MARK.LT..75) GO TO 6
C     NOTE WE TAKE OUT THE UPPER 15TH BIT TO AVOID
C     INTEGER OVERFLOW IN TESTING FOR THE LOWER EIGHT
C     BITS.  THIS IS NOT NECESSARY WHEN
C     WE DIVIDE BY 256 TO OBTAIN THE UPPER BITS.
      X=YMIMR(MARK+(LEVEL-1)*LTH)
      IF(X.GT.32767.0)X=X-32768.
      DCT2(IND)=DCT1(IAND(IFIX(X),255)+1)
      IORDR(IND)=IAND(IFIX(X),255)+1
      IND=IND-1
      ISKIP=IAND(IFIX((YMIMR(MARK+(LEVEL-1)*LTH))/256),255)+1
      GO TO 7
6     CONTINUE
      RETURN
      END
```

```
C        PROGRAM NAME:SORTN.FTN  ORIGINATED:13-SEP-79
C                                UPDATED:13-SEP-79
C        PERFORMS SORT OF DCT1 INTO DCT2 IN MAX-TO-MIN ORDER
C        W/ INDEX ORDER RETURNED IN IORDR
C
C        SUBROUTINE OVR7
         SUBROUTINE SORTN(LTH)
         COMMON/SORT/DCT1(256),DCT2(256),IORDR(256),XR(512),XI(512)
         COMMON/BLOCK1/INIT1,DCT(256),LTH,NSTAGE
         DO 202 I=1,LTH
         DCT2(I)=DCT1(I)
202      IORDR(I)=I
1015     LTHM1=LTH-1
         DO 100 J=1,LTHM1
         JP1=J+1
         IF(DCT2(J).GE.DCT2(JP1))GO TO 100
         TEMP=DCT2(J)
         DCT2(J)=DCT2(JP1)
         DCT2(JP1)=TEMP
         ITEMP=IORDR(J)
         IORDR(J)=IORDR(JP1)
         IORDR(JP1)=ITEMP
         GO TO 1015
100      CONTINUE
         RETURN
         END
```

```
      SUBROUTINE OVR8(NBPF,NBRPF)
C     SUBROUTINE DITBA(NBPF)
C     ROUTINE DITBA IS DIGITAL TO BINARY CONVERSION
C     ROUTINE
C     NBPF IS THE NUMBER OF BITS PER FRAME
      COMMON/BLOCK1/INIT1,DCT(256),LTH,NSTAGE
      COMMON/SORT/DCT1(256),DCT2(256),IORDR(256),XR(512),XI(512)
      COMMON/DITBA/ QTDCT(256),QTPAR(8),QTDC,QTVAR,QTM,QTG
      COMMON/BLOCK6/IBIT(256),IPDEC,INBA(500)
      DIMENSION INB(6)
      INTEGER QTDCT,QTPAR,QTDC,QTVAR,QTM,QTG
D     TYPE 57,NBRPF,IPDEC
D57   FORMAT(1X,'NBRPF=',I4,3X,'IPDEC=',I4)
      DO 10 I=1,NBRPF
10    INBA(I)=0
C     XI ARRAY CONTAINS THE PROTECTED BITS OF DCT
C     NBPT IS NUMBER OF BITS OF DCT COEFFICIENTS TO BE PROTECTED
      NBPT=3*45-3-34
      IF(IPDEC.EQ.1)NBPT=NBPT-8
C     SELECT THE DCT BITS TO BE PROTECTED
      INP=0
      IBST=IBIT(1)
D15   FORMAT(1X,'IT1=',I3,1X)
110   CONTINUE
      DO 100 I=1,LTH
      IAB=IBIT(I)
D     WRITE(4,16)I,IBST,IAB
D16   FORMAT(1X,'I=',I3,2X,'IBST=',I3,2X,'IAB=',I3,2X)
      IF(IAB.LT.IBST)GO TO 130
      IF(IAB.EQ.0)GO TO 100
      INP=INP+1
      IT1=2**(IAB-1)
D     WRITE(4,15)IT1
      IT2=IT1-1
      IT3=QTDCT(I)
      QTDCT(I)=IAND(IT3,IT2)
      IBIT(I)=IBIT(I)-1
      II=IT3/IT1
      XI(INP)=FLOAT(II)+0.1
      IF(INP.GE.NBPT)GO TO 120
100   CONTINUE
130   IBST=IBST-1
      GO TO 110
120   CONTINUE
C
C     CONVERT PARCOR INTO BICNARY VECTOR
C     AND LOAD INTO INBA
C     INB IS THE TEMPORARY BINARY VECTOR
C     ALLOCATE 5,5,4,4,3,3,2,2 BITS TO THE PARCORS
      NBA=0
      DO 20 I=1,8
      IDT=6-(I+1)/2
      CALL DBCONV(QTPAR(I),IDT,INB)
      DO 30 J=1,IDT
      NBA=NBA+1
30    INBA(NBA)=INB(J)
20    CONTINUE
C     CONVERT VARIANCE INTO BINARY VECTOR
      CALL DBCONV(QTVAR,5,INB)
      DO 40 J=1,5
      NBA=NBA+1
40    INBA(NBA)=INB(J)
C     NOW CONVERT VOICED INTO BINARY VECTOR
      NBA=NBA+1
      INBA(NBA)=IPDEC
C     PROTYECT 3 BITS FROM DCBIAS
      IBDC=QTDC
      DO 41 I=1,3
      IPD=5-I
      IT1=2**IPD
      IT2=IT1-1
      II=IBDC/IT1
      IBDC=IAND(IBDC,IT2)
      NBA=NBA+1
      INBA(NBA)=II
41    CONTINUE
      QTDC=IBDC
C     NOW GENERATE BINARY PITCH AND PITCH GAIN
      IF(IPDEC.EQ.0)GO TO 200
      CALL DBCONV(QTM,6,INB)
      DO 210 J=1,6
      NBA=NBA+1
210   INBA(NBA)=INB(J)
      CALL DBCONV(QTG,2,INB)
      INBA(NBA+1)=INB(1)
      INBA(NBA+2)=INB(2)
      NBA=NBA+2
200   CONTINUE
C
C     NOW LOAD XI INTO THE BINARY ARRAY
C
      ISTP=1
      IEDP=45-NBA
      IEDP1=IEDP
      DO 310 NEB=1,3
      IF(IEDP.GT.0) GO TO 400
      IEDP=45
      NBA=63
      IF(IEDP1.EQ.0)GO TO 310
      INBA(64)=INBA(46)
      INBA(46)=0
      NBA=64
      IEDP=44
      IF(IEDP1.EQ.-1)GO TO 310
      INBA(65)=INBA(47)
      INBA(47)=0
      NBA=65
      IEDP=43
      GO TO 310
400   CONTINUE
      DO 300 I=ISTP,IEDP
      NBA=NBA+1
300   INBA(NBA)=XI(I)
      NBA=NEB*63
      ISTP=IEDP+1
      IEDP=IEDP+45
310   CONTINUE
510   CONTINUE
C     LOAD THE REMAINING TWO BITS OF THE DCBIAS
      CALL DBCONV(QTDC,2,INB)
      DO 599 J=1,2
      NBA=NBA+1
599   INBA(NBA)=INB(J)
C
C     NOW LOAD THE REMAINING THE UNPROTECTWED
C     BITS FROM THE QTDCT
      DO 500 I=1,LTH
      IDT=IBIT(I)
      IF(IDT.LE.0)GO TO 500
```

```
        CALL DBCONV(QTDCT(I),IDT,INB)
        DO 600 J=1,IDT
        NBA=NBA+1
500     INBA(NBA)=INB(J)
600     CONTINUE
)       TYPE 808,NBA
D808    FORMAT(1X,'NBAIN SER. ROUTINE=',I4)
D       WRITE(4,909)(I,INBA(I),I=1,NBRPF)
D909    FORMAT(/1X,8(2X,I3,2X,'=',2X,I3))
        RETURN
        END
C
C       DBCONVERT TAKES A DECIMAL NUMBER AND RETURNS
C       ITS BINARY EQUIVALENT
C       SUBROUTINE DBCONV(IX,LIB,INB)
C       IX IS THE DECIMAL NUMBER
C       LIB IS THE NUMBER OF BITS TO BE ALLOCATED
C       INB IS THE BINARY VECTOR EQUIVALENT
        SUBROUTINE DBCONV(IX,LIB,INB)
        DIMENSION INB(1)
        IY=IX
        DO 10 I=1,LIB
        IR=LIB+1-I
        INB(IR)=MOD(IY,2)
10      IY=IY/2
        RETURN
        END
```

```
C       ENCBCH.FTN
C
C       ENCODING OF A (63,45)BCH CODE
        SUBROUTINE OVR9
C       SUBROUTINE ENCBCH
        COMMON/DECBCH/NES,KT
        COMMUN/BLOCK6/IBIT(256),IPDEC,INBA(500)
        DIMENSION INC(26),ING(19)
        DATA ING/1,1,1,1,0,0,0,0,0,1,0,1,1,0,0,1,1,1,1/
C       CALCULATE PARITY BITS
        DO 10 I=1,63
        KTI=(KT-1)*63+I
10      IBIT(I)-INBA(KTI)
        CALL GF2DIV(IBIT,63,ING,19,INC,NC)
C       STORE PARITY BITS
        DO 20 I=1,NC
        KTI=(KT-1)*63+I+45
20      INBA(KTI)=INC(I)
        RETURN
        END
```

```
      SUBROUTINE OVR10
C     ED3BCH.FTN
C     ENCODING, DECODING TEST OF BCH CODE
C     MARCH 20, 1979
C     LBCH;LENGTH OF BCH CODE
C     POLYNOMIALS ARE ORDERED IN DESCENDING POWER SERIES
C     LBCH=2**MBCH-1
C     THIS ROUTINE CORRECT 3 ERRORS
      COMMON/BLOCK6/INA(256),IPDEC,INBA(500)
      COMMON/DECBCH/NES,KT
      COMMON/BLOC30/ICOEF1(64),ICOEF3(64),ICOEF5(64)
      COMMON/BLOC31/ICOEFF(64),IORDR(64)
      DIMENSION NERL(6),INB(7),INC(7)
      DIMENSION ISD1(9),ISD3(9),ISD5(9)
C
C     READ INBA VECTOR INTO INA VECTOR IN BLOCK OF 63
      DO 10 I=1,63
      KTI=(KT-1)*63+I
10    INA(I)=INBA(KTI)
C
C     DECODING ROUTINE
C
C     ******************************/****
C
C     FIRST SET UP ALL OF THE COEFFICIENT
C      AND POWER TABLES TO BE USED BY THE SUBROUTINES
C     IF WE HAVE ALREADY GENERATED THE TABLES FOR THE
C     DECODING ROUTINE DO NOT DO IT AGAIN
      IF(KT.GE.2)GO TO 1111
      MBCH=6
      LBCH=2**MBCH-1
      CALL GENTAB(MBCH)
C
C     CALCULATE POWER SUMS
C     R(ALPHA**1),R(ALPHA**3),R(ALPHA**5)
1111  CONTINUE
      CALL GF2POL(MBCH,INA,ISD1,IP1,
     2ISD3,IP3,ISD5,IP5)
      IPSIG1=IP1
D     WRITE(5,101)(ISD1(I),I=1,MBCH)
D     WRITE(5,202)(ISD3(I),I=1,MBCH)
D     WRITE(5,303)(ISD5(I),I=1,MBCH)
101   FORMAT(1X,'S1=',18I1)
202   FORMAT(1X,'S3=',18I1)
303   FORMAT(1X,'S5=',18I1)
D     WRITE(5,102)IP1,IP3,IP5
102   FORMAT(I6,1X,I6,1X,I6)
C     CHECK ERROR RANGE
      IF(IP1.EQ.-1.AND.IP3.EQ.-1.AND.IP5.EQ.-1)GO TO 1599
      GO TO 50
C     NO CHANNEL ERROR
1599  CONTINUE
D     WRITE(5,1600)
1600  FORMAT(1X,'NO CHANNEL ERROR')
      RETURN
50    CONTINUE
C     CORRECT CHANNEL ERROR
C
C     CALCULATE SIGMA(I),I=1,3
C     CALCULATE DET(3),I.E.,S1**3+S3
      CALL MODPOW(IP1,3,LBCH,I3MOP1)
      CALL INVLOK(MBCH,I3MOP1,INA)
      CALL GF2ADD(INA,MBCH,ISD3,MBCH,INA,ND)
      CALL LOOKUP(MBCH,INA,IPDENO)
```

```
C       IF(ISW0.EQ.2)WRITE(5,303)(INB(I),I=1,6)
        NES=3
        IF(IPDENO.GT.-1) GO TO 70
        NES=1
C       ONLY ONE ERROR OCCUR
        GO TO 80
70      CONTINUE
C       CALCULATE SIGMA(2) AND SIGMA(3)
        CALL MODPOW(IP1,2,LBCH,INUMOP)
        CALL MODMUL(INUMOP,IP3,LBCH,INUMOP)
        CALL INVLOK(MBCH,INUMOP,INB)
        CALL GF2ADD(INB,MBCH,ISD5,MBCH,ISD3,NC)
        CALL LOOKUP(MBCH,ISD3,IPSIG2)
C       IF(ISW0.EQ.2)WRITE(5,303)(INC(I),I=1,MBCH)
        CALL MODDIV(IPSIG2,IPDENO,LBCH,IPSIG2)
        CALL INVLOK(MBCH,IPSIG2,ISD3)
C       WRITE(5,606)(ISD3(I),I=1,6)
C606    FORMAT(1X,'SIGMA2=',18I1)
C       IF(ISW0.EQ.2.AND.ISW1.EQ.1)WRITE(5,404)I,(ISD3(J),J=1,MBCH)
C404    FORMAT(1X,'N=',I3,3X,8I1)
C
C       IF(ISW0.EQ.2)WRITE(5,202)(ISD3(I),I=1,MBCH)
C       CALCULATE SIGMA(3)
        CALL MODMUL(IPSIG2,IP1,LBCH,IPSIG3)
        CALL INVLOK(MBCH,IPSIG3,INB)
        CALL GF2ADD(INA,MBCH,INB,MBCH,ISD5,NC)
        CALL LOOKUP(MBCH,ISD5,IPSIG3)
C       WRITE(5,707)(ISD5(I),I=1,6)
C707    FORMAT(1X,'SIGMA3=',18I1)
        IF(IPSIG3.EQ.-1)NES=2
80      CONTINUE
D       TYPE 505,NES
505     FORMAT(5X,'NUMBER OF ERRORS =',I3)
C       CORRECT NES ERROR BY CHIEN'S SEARCH METHOD
C
C       IPSIG1,IPSIG2,IPSIG3 ARE THE EXPONENTS OF SIGMA1,SIGMA2,SIGMA3
C       ISD1,ISD3,ISD5, ARE THE VECTOR POWER SUMS 1,3,5
        NEST=0
        DO 11 II=1,LBCH
        II1=II-1
        IF(II1.EQ.0)II1=LBCH
        IF(NES.EQ.1.AND.IPSIG1.EQ.0) GO TO 33
        IF(NES.EQ.1)GO TO 44
        CALL GF2ADD(ISD1,MBCH,ISD3,MBCH,INB,NC)
        CALL GF2ADD(ISD5,MBCH,INB,MBCH,INC,NC)
C       IF(ISW0.EQ.2)WRITE(5,303)(INC(I),I=1,6)
        CALL LOOKUP(MBCH,INC,IPSUM)
C       IF(ISW0.EQ.2.AND.ISW1.EQ.1)WRITE(5,303)(INC(I),I=1,MBCH)
C       IF(INC(I).EQ.1)GO TO 44
        IF(IPSUM.NE.0) GO TO 44
C       CORRECT ERROR
33      CONTINUE
        KTT=(KT-1)*63+II1
        NEST=NEST+1
        NERL(NEST)=KTT
C       WRITE(5,1700)II1
C1700   FORMAT(10X,'CORRECTED ERROR LOCATION=',I3)
44      CONTINUE
C       SHIFT ISV1,ISV3,ISV5
C       FIRST MULTIPLY SIGMA1 BY ALPHA
        CALL MODMUL(IPSIG1,1,LBCH,IPSIG1)
        CALL INVLOK(MBCH,IPSIG1,ISD1)
34      IF(NES.EQ.1)GO TO 11
C       MULTIPLY SIGMA2 BY ALPHA SQUARED
```

```
      CALL MODMUL(IPSIG2,2,LBCH,IPSIG2)
      CALL INVLOK(MBCH,IPSIG2,ISD3)
C     MULTIPLY SIGMA3 BY ALPHA CUBED
C     TYPE 808,IPSIG3
C808  FORMAT(5X,'IPSIG3=',I3)
      CALL MODMUL(IPSIG3,3,LBCH,IPSIG3)
36    CALL INVLOK(MBCH,IPSIG3,ISD5)
C     WRITE(5,707)(ISD5(I),I=1,6)
11    CONTINUE
C     CHECK ERROR STATUS
C     TYPE 506,NES,NEST
506   FORMAT(5X,'NES=',I3,3X,'NEST=',I3)
      IF(NES.EQ.NEST)GO TO 888
      NES=4
      RETURN
C     CORRECT ERRORS
888   CONTINUE
      DO 72 I=1,NES
      KTT=NERL(I)
D     TYPE 5060,KTT
5060  FORMAT(5X,'ERROR LOCATION =',I5)
C     TYPE 507,INBA(KTT)
C507  FORMAT(5X,'OLD INBA VALUE BEFORE CORRECTIONS=',I3)
      INBA(KTT)=IEOR(INBA(KTT),1)
C     TYPE 508,INBA(KTT)
C508  FORMAT(5X,'INBA VALUE AFTER CORRECTION=',I3)
72    CONTINUE
      RETURN
      END
      SUBROUTINE MODPOW(IEXP1,MULT,IMOD,IEXP2)
C     EXP1 IS THE EXPONENT TO BE MULTIPLIED
C     MULT IS THE MULTIPLIER OF THE EXPONENT
C     IMOD IS THE INTEGER THAT IT IS ALL MODULOED TO
C     EXP2 IS THE RETURNED EXPONENT
      ITEMP=MOD((MULT*IEXP1),IMOD)
      IF(IEXP1.EQ.-1) ITEMP=-1
      IEXP2=ITEMP
      RETURN
      END
      SUBROUTINE MODMUL(IEXP1,IEXP2,IMOD,IEXP3)
C     EXP1 AND EXP2 WILL BE ADDED AND MODULOED BY IMOD
      ITEMP=MOD((IEXP1+IEXP2),IMOD)
      IF(IEXP1.EQ.-1.OR.IEXP2.EQ.-1)ITEMP=-1
      IEXP3=ITEMP
C     TYPE5,IEXP1,IEXP2,IEXP3
C5    FORMAT(2X,I5,I5,I5)
      RETURN
      END
      SUBROUTINE MODDIV(IEXP1,IEXP2,IMOD,IEXP3)
C     IEXP2 WILL BE SUBTRACTED FROM IEXP1 AND MODULOED IMOD
      ITEMP=MOD((IEXP1-IEXP2+IMOD),IMOD)
      IF(IEXP1.EQ.-1) ITEMP=-1
      IEXP3=ITEMP
      RETURN
      END
      SUBROUTINE UNPACK(MBCH,ISUM,IVEC)
      DIMENSION  IVEC(1)
      DO 10 J=1,MBCH
      INORM=2**(MBCH-J)
      ISHIFT=(ISUM/INORM)
      IT=IAND(1,ISHIFT)
      IVEC(J)=IT
27    FORMAT(1X,'IBIT=',I2)
10    CONTINUE
C     WRITE(5,25)ISUM
```

```
25      FORMAT(1X,'SUM=',I4)
        WRITE(5,26)(IVEC(J),J=1,MBCH)
26      FORMAT(1X,'VECTOR=',6(I1,1X))
        RETURN
        END
```

```
      SUBROUTINE GF2POL(MBCH,INA,ISD1,IEXP1,
     2ISD3,IEXP3,ISD5,IEXP5)
      DIMENSION ISD3(1),ISD5(1),ISD1(1),INA(1)
C     IORD  DETERMINES WHICH OF THE POWER SUMS IS GENERATED,
C     S(1),S(3),S(5),...
C     ISD   IS THE BINARY VALUED VECTOR CORRESPONDING TO THE
C     GF(2)FIELD
C     IEXP  IS THE EXPONENT CORRESPONDING TO THE VECTOR IN THE FIELD
      IBLOCK=(2**MBCH)-1
      IF(INA(1).EQ.0) GO TO 12
      IPOWR1=1
      IPOWR3=3
      IPOWR5=5
      CALL  INVLOK(MBCH,IPOWR1,ISD1)
      CALL  INVLOK(MBCH,IPOWR3,ISD3)
      CALL  INVLOK(MBCH,IPOWR5,ISD5)
      ISD1(MBCH)=IEOR(INA(2),ISD1(MBCH))
      ISD3(MBCH)=IEOR(INA(2),ISD3(MBCH))
      ISD5(MBCH)=IEOR(INA(2),ISD5(MBCH))
      GO TO 19
12    DO 13 I=1,MBCH-1
      ISD1(I)=0
      ISD3(I)=0
      ISD5(I)=0
13    CONTINUE
      ISD1(MBCH)=INA(2)
      ISD3(MBCH)=INA(2)
      ISD5(MBCH)=INA(2)
19    DO 20 I=2,IBLOCK-1
      CALL CHARTS(MBCH,ISD1,ISD3,ISD5)
      ISD1(MBCH)=IEOR(INA(I+1),ISD1(MBCH))
      ISD3(MBCH)=IEOR(INA(I+1),ISD3(MBCH))
      ISD5(MBCH)=IEOR(INA(I+1),ISD5(MBCH))
30    CONTINUE
C     BEFORE YOU GO GET ITS POWER
      CALL LOOKUP(MBCH,ISD1,IEXP1)
      CALL LOOKUP(MBCH,ISD3,IEXP3)
      CALL LOOKUP(MBCH,ISD5,IEXP5)
      RETURN
      END
      SUBROUTINE LOOKUP(MBCH,IVEC,IEXP)
C     THIS SUBROUTINE RETURNS THE POWER OF THE ELEMENT
C     IN THE GF(2) FIELD CORRESPONDING TO ITS BINARY VALUED VECTOR
C     NOTICE THAT THE ZEROED VALUE VECTOR TRAPS TO THE EXPONENT -1
C     IVEC IS THE BINARY VALUED VECTOR
C     IEXP IS THE POWER OF THE FIELD ELEMENT
      COMMON/BLOC31/ICOEFF(64),IPOWER(64)
      DIMENSION  IVEC(1)
3     INDEX=0
      DO 10 I=1,MBCH
      INDEX=((2**(MBCH-I))*IVEC(I))+INDEX
10    CONTINUE
      IEXP=IPOWER(INDEX+1)
      RETURN
      END
      SUBROUTINE INVLOK(MBCH,IPOWER,IVEC)
C     SUBROUTINE INVLOK RETURNS THE BINARY VALUED VECTOR IVEC
C     FROM THE GF2 FIELD CORRESPONDING TO THE POWER OF THE ELEMENT OF THE FIELD
C     IPOWER  DETERMINES WHICH OF THE 6 BIT VECTORS IS INDEXED
C     IVEC  IS THE VECTOR FROM THE FIELD
      COMMON/BLOC31/ ICOEFF(64),IEXP(64)
      DIMENSION  IVEC(9)
      ITEMP=ICOEFF(IPOWER+1+1)
      CALL UNPACK(MBCH,ITEMP,IVEC)
C9    DO 10 I=1,MBCH
C     IVEC(I)=ICOEFF((MBCH*(IPOWER+1))+I)
C10   CONTINUE
      RETURN
      END
      SUBROUTINE CHARTS(MBCH,ISD1,ISD3,ISDE)
C     THIS SUBROUTINE ROUTINE THE APPROPRIATELY ALTERED
C     VECTOR TO THE POWER SUM
C     ISD IS JUST IVEC*(ALPHA**ITREE).
      COMMON/BLOC30/ICOEF1(64),ICOEF3(64),ICOEF5(64)
      COMMON/BLOC31/ICOEFF(64),IPOWER(64)
      DIMENSION ISD1(1),ISD3(1),ISD5(1)
      INDEX1=0
      INDEX3=0
      INDEX5=0
      ICOUNT=2**MBCH
3     DO 10 I=1,MBCH
      INDEX1=((2**(MBCH-I))*ISD1(I))+INDEX1
      INDEX3=((2**(MBCH-I))*ISD3(I))+INDEX3
      INDEX5=((2**(MBCH-I))*ISD5(I))+INDEX5
10    CONTINUE
      ITEMP1=ICOEF1(INDEX1+1)
      ITEMP3=ICOEF3(INDEX3+1)
      ITEMP5=ICOEF5(INDEX5+1)
      CALL UNPACK(MBCH,ITEMP1,ISD1)
      CALL UNPACK(MBCH,ITEMP3,ISD3)
      CALL UNPACK(MBCH,ITEMP5,ISD5)
C     DO 51 I=1,MBCH
C     ISD5(I)=ICOEF5((MBCH*(INDEX5))+I)
C51   CONTINUE
C41   DO 52 I=1,MBCH
C     ISD1(I)=ICOEF1((MBCH*(INDEX1))+I)
C52   CONTINUE
C43   DO 54I=1,MBCH
C     ISD3(I)=ICOEF3((MBCH*(INDEX3))+I)
C54   CONTINUE
      RETURN
      END
      SUBROUTINE GENTAB(MBCH)
      COMMON/BLOC30/ICOEF1(64),ICOEF3(64),ICOEF5(64)
      COMMON/BLOC31/ICOEFF(64),IORDR(64)
      DIMENSION  INA(127),INC(8),INB(8),IPOLY(21)
      DIMENSION INDEX(3),IVEC(1),ISUM(64),ITEMP(64)
C     DIMENSION ICOEF2(596),ICOEF4(596)
      DATA IPOLY/1,0,0,1,0,1,1,0,0,0,0,1,1,1,0,0,0,1,0,0,1/
      DATA INDEX/0,6,13/
      ICNT=1
      ICOUNT=2**MBCH
      ICOUN2=ICOUNT-1
      MBC2=MBCH+1
C     IF(INIT.GT.1) GO TO 30
      IPT=MBCH-5
      DO 10 I=1,MBCH+1
      INB(I)=IPOLY(INDEX((IPT+1))+I)
10    CONTINUE
C     WRITE(5,169)(INB(I),I=1,MBCH+1)
C169  FORMAT(1X,'INB=',18I1)
C
C     NOW WE WILL PACK THE BITS INTO ONE WORD
C
C     ICOEFF(1)=0
C
C     DO 14 M=1,MBCH
C14   ICOEFF(M)=0
      ISUM(ICNT)=0
      DO 16 K=1,ICOUNT-1
      ICNT=ICNT+1
      ISUM(ICNT)=0
      DO 11 I=1,ICOUNT-1
      INA(I)=0
11    CONTINUE
      INA(ICOUNT-K)=1
C     WRITE(5,170)(INA(I),I=1,ICOUNT-1)
C170  FORMAT(1X,'INA=',70I1)
      CALL GF2DIV(INA,ICOUN2,INB,MBC2,INC,NC)
C     WRITE(5,171)(INC(I),I=1,NC)
C171  FORMAT(1X,'INC=',18I1)
C
C     WE WILL PACK THE BITS INTO ONE INTEGER WORD
C     THE VALUE WILL BE STORED AS THE SUM OF THE MBCH BITS
C
      DO 12 J=1,MBCH
C     ICOEFF((K*MBCH)+J)=INC(J)
      ISUM(ICNT)=ISUM(ICNT)+(INC(J)*(2**(MBCH-J)))
      ICOEFF(ICNT)=ISUM(ICNT)
12    CONTINUE
16    CONTINUE
C     WRITE(5,174)(ISUM(I),I=1,ICOUNT)
C174  FORMAT(1X,'DECIMAL SUM OF INCREASING ALPHA VECTOR=',16I3)
C     WRITE(5,172)(ICOEFF(I),I=1,(MBCH*ICOUNT))
C172  FORMAT(1X,'ICOEFF=',70I1)
C
C     NOW ORDER THE ALPHA LEVELS
C     IORD WILL CONTAIN THE LOOKUP CHART
C
      DO 202 I=1,ICOUNT
202   IORDR(I)=I-2
1015  LTHM1=ICOUNT-1
      DO 100 J=1,LTHM1
      IF(ISUM(J).LE.ISUM(J+1))GO TO 100
      ITEMP1=ISUM(J)
      ISUM(J)=ISUM(J+1)
      ISUM(J+1)=ITEMP1
      ITEMP2=IORDR(J)
      IORDR(J)=IORDR(J+1)
      IORDR(J+1)=ITEMP2
      GO TO 1015
100   CONTINUE
C     WRITE(5,910)(I,IORDR(I),I=1,ICOUNT)
C910  FORMAT(1X,6(1X,'IORDR(',I3,')=',I3,2X))
C     WRITE(5,909)(I,ISUM(I),I=1,ICOUNT)
C909  FORMAT(1X,4(1X,'ISUM(',I3,')=',I3,2X))
C
C173  FORMAT(1X,'IVEC=',18I1)
C     THIS SUBROUTINE GENERATES FIVE DIFFERENT CHARTS
C     THE CHARTS ARE APPROPRIATELY WEIGHTED BY ALPHA**J.
C     THE SUM OF THE VECTOR CORRESPONDING TO THIS WEIGHTING IS STORED
C     IN THE CHART AND IS THEN UNPACKED INTO ITS MBCH VECTOR AT
C     A LATER TIME.
C
      INDEX1=0
      INDEX3=0
      INDEX5=0
      ICOUNT=2**MBCH
      DO 80J1=1,3
      J=(J1-1)*2+1
      DO 17 ISUM2=1,ICOUNT
      ISUM1=ISUM2-1
      IEXP1=IORDR(ISUM1+1)
      CALL MODMUL(IEXP1,J,ICOUNT-1,IEXP2)
      DO 13 I=1,MBCH
C
C     INSTEAD LET'S STORE THE SUM
C     ITEMP(ISUM2)=ICOEFF(IEXP2+2)
      ITEMP((ISUM1*MBCH)+I)=ICOEFF((MBCH*(IEXP2+1))+I)
13    CONTINUE
17    CONTINUE
      IF(J.EQ.1)GO TO 21
C     IF(J.EQ.2)GO TO 22
      IF(J.EQ.3)GO TO 23
C     IF(J.EQ.4)GO TO 24
C     DO 38 I=1,ICOUNT
      DO 38 I=1,ICOUNT*MBCH
      ICOEF5(I)=ITEMP(I)
38    CONTINUE
      GO TO 15
21    DO 34 I=1,ICOUNT
C     DO 34 I=1,ICOUNT*MBCH
      ICOEF1(I)=ITEMP(I)
34    CONTINUE
      GO TO 15
C22   DO 35 I=1,ICOUNT
C     DO 35 I=1,ICOUNT*MBCH
C     ICOEF2(I)=ITEMP(I)
C35   CONTINUE
C     GO TO 15
23    DO 36 I=1,ICOUNT
C     DO 36 I=1,ICOUNT*MBCH
      ICOEF3(I)=ITEMP(I)
36    CONTINUE
      GO TO 15
C24   DO 37 I=1,ICOUNT
C     DO 37 I=1,ICOUNT*MBCH
C     ICOEF4(I)=ITEMP(I)
C37   CONTINUE
15    CONTINUE
80    CONTINUE
```

```
RETURN
END
```

```
C       BATSD.FTN
C       MARCH 13, 1979
C       CONVERT INPUT BINARY VECTOR INTO DECIMAL SIDE INFORMATION
C       SUBROUTINE OVR11
        SUBROUTINE BATSD(NEPB)
        COMMON/DITBA/QTDCT(256),QTPAR(8),QTDC,QTVAR,QTM,QTG
        COMMON/BLOCK6/IBIT(256),IPDEC,INBA(500)
        DIMENSION INB(6)
        INTEGER QTDCT,QTPAR,QTDC,QTVAR,QTM,QTG
C       READ PARCOR(I),I=1,8
        NUV=IPDEC
        NEPA=3
        NBA=0
        DO 10 I=1,8
        IDT=6-(I+1)/2
        DO 20 J=1,IDT
        NBA=NBA+1
 20     INB(J)=INBA(NBA)
        CALL BDCONV(INB,IDT,QTPAR(I))
 10     CONTINUE
C       READ QTVAR
        DO 30 J=1,5
        NBA=NBA+1
 30     INB(J)=INBA(NBA)
        CALL BDCONV(INB,5,QTVAR)
C       READ V/UV
        NBA=NBA+1
        IPDEC=INBA(NBA)
C       READ DCBIAS
        QTDC=0
        IF(NEPB.LE.0)GO TO 60
        DO 40 I=1,NEPB
        IT=5-I
        ITT=2**IT
        NBA=NBA+1
        QTDC=QTDC+INBA(NBA)*ITT
 40     CONTINUE
 60     CONTINUE
C       ADD REMAINDER OF DCBIAS
        NDCR=5-NEPB
        IF(NDCR.EQ.0)GO TO 80
        NDCPC=NEPB*63
        IF(NEPB.EQ.0.AND.NUV.EQ.0)NDCPC=34
        IF(NEPB.EQ.0.AND.NUV.EQ.1)NDCPC=42
        DO 70 I=1,NDCR
        IP=NDCPC+I
        INB(I)=INBA(IP)
 70     CONTINUE
        CALL BDCONV(INB,NDCR,IBPG)
        QTDC=QTDC+IBPG
 80     CONTINUE
        IF(IPDEC.EQ.0)RETURN
        DO 50 J=1,6
        NBA=NBA+1
 50     INB(J)=INBA(NBA)
        CALL BDCONV(INB,6,QTM)
        NBA=NBA+1
        INB(1)=INBA(NBA)
        IF(NBA.GT.45)INB(1)=INBA(NBA+18)
        NBA=NBA+1
        INB(2)=INBA(NBA)
        IF(NBA.GT.45)INB(2)=INBA(NBA+18)
        CALL BDCONV(INB,2,QTG)
```

```
      END
C     SUBROUTINE BDCONV TO TAKE A BINARY VECTOR TO A DECIMAL NUMBER
      SUBROUTINE BDCONV(INB,LIB,IY)
      DIMENSION INB(1)
      IY=0
      IF(LIB.LE.0)RETURN
      DO 10 I=1,LIB
      IT=2**(I-1)
10    IY=IY+INB(LIB+1-I)*IT
      RETURN
      END
```

```
C       SBADCT.FTN
C       THIS RETURNS THE DECIMAL VALUE OF THE DCT COEFFICIENTS
C       FROM THE CODED AND (PACKED BINARY VECTOR INBA
        SUBROUTINE OVP12(NBPF)
C       SUBROUTINE SBADCT(NBPF)
        COMMON/SORT/DCT1(256),DCT2(256),IORDR(256),XR(512),XI(512)
        COMMON/BLOCK1/INIT1,DCT(256),LTH,NSTAGE
        COMMON/BLOCK6/IBIT(256),IPDEC,INBA(500)
        DIMENSION INB(6)
C       IBIT(I),I=1,LTH ARE BIT ASSIGNMENTS VECTOR
C       DCT(I),I=1,LTH WILL BE DCT QUANTIZER LEVEL IN REAL FORMAT.
C       THEN THE DCT WIL BE INTEGERIZED AND PASSED TO ORDCT
        NEPB=3
C       NBRPF IS THE TOTAL NUMBER OF BITS PER FRAME PLUS THE PARITY BITS
        NBRPF=NBPF+3*18
D       TYPE 707,NBPF,NBRPF
D707    FORMAT(1X,'NBPF=',I4,3X,'NBRPF=',I4)
D       WRITE(4,808)(I,INBA(I),I=1,NBPPF)
D808    FORMAT(/1X,8(2X,I3,2X,'=',2X,I3))
        NUV=IPDEC
        NBA=34+NEPB
        IF(NUV.EQ.1)NBA=NBA+8
        NBPT=3*45-3-34
        IF(NUV.EQ.1)NBPT=NBPT-8
C       INI XI(I),DCT(I),I=1,LTH
        DO 10 I=1,LTH
        DCT(I)=0.1
10      XI(I)=FLOAT(IBIT(I))+0.1
        IF(NEPB.LE.0)GO TO 120
C       REMOVE REDUNDANT BIT FROM INBA(I) VECTOR
        NF1=NBRPF
        DO 40 J=1,3
        NF1=NF1-18
        NI1=J*45+1
        DO 20 I=NI1,NF1
20      INBA(I)=INBA(I+18)
40      CONTINUE
D       WRITE(4,909)(I,INBA(I),I=1,NF1)
D909    FORMAT(/1X,8(2X,I3,2X,'=',2X,I3))
C       FIND NEW BIT ASSIGMENTS AND READ SCRAMBLED DATA
        INP=0
        IBST=IBIT(1)
110     CONTINUE
        DO 100 I=1,LTH
        IAB=IBIT(I)
        IF(IAB.LT.IBST)GO TO 130
        IF(IAB.EQ.0)GO TO 100
        INP=INP+1
        NBA=NBA+1
        ITT=2**(IAB-1)
        IBIT(I)=IBIT(I)-1
        DCT(IORDR(I))=DCT(IORDR(I))+FLOAT(INBA(NBA)*ITT)
        IF(INP.GE.NBPT)GO TO 120
100     CONTINUE
130     IBST=IBST-1
        GO TO 110
120     CONTINUE
C       READ DCBIAS
C       NBA=90
C       REMEMBER TO MOVE NBA POINTER 2 BITS TO ACCOUNT
C       FOR THE DC BIAS BITS YOU ALREADY PLUCCKED OUT
C       NBA=NBA+2
C       IBIT(I),I=1,LTH ARE REDUCED BIT ASSIGNMENT
C       READ DCT(I),I=1,LTH FROM INBA AND XI
```

```
C       CONVERT INBA INTO DCT
        DO 500 I=1,LTH
        IDT=IBIT(I)
        IF(IDT.LE.0)GO TO 500
C       READ INPUT VECTOR
        DO 600 J=1,IDT
        NBA=NBA+1
600     INB(J)=INBA(NBA)
        CALL BDCONV(INB,IDT,IY)
        DCT(IORDR(I))=DCT(IORDR(I))+FLOAT(IY)
500     CONTINUE
C       TYPE911,NBA
C911    FORMAT(1X,'NBA IN DESER. ROUTINE =',I4)
C       NOW SORT DCT INTO AN ORDERED DCT2 ARRAY\
        DO 505 I=1,LTH
505     DCT2(I)=DCT(IORDR(I))
        DO 510 I=1,LTH
510     IBIT(I)=IFIX(X1(I))
        RETURN
        END
        SUBROUTINE BDCONV(INB,LIB,IY)
        DIMENSION INB(1)
        IY=0
        IF(LIB.LE.0)RETURN
        DO 10 I=1,LIB
        IT=2**(I-1)
10      IY=IY+INB(LIB+1-I)*IT
        RETURN
        END
```

```
C       MARCH 16, 1979
C       CHANNEL ERROR SIMULATION ROUTINE
        SUBROUTINE OVR13(NBRPF,PROB.IRN.JRN.NERB,NEPB)
C       SUBROUTINE CEIR(INBA,NBRPF,PROB.IRN,JRN,NERB,NEPB)
        COMMON/BLOCK6/IBIT(256),IPDEC,INBA(500)
        COMMON/SW/ICOUNT,IPRSW
        DIMENSION NERB(6)
C       TYPE 808,NBRPF,NEPB
808     FORMAT(1X,I4,2X,I4)
C       WRITE(5,809)(INBA(I),I=1,NBRPF)
809     FORMAT(1X,63(I1))
        NEPB1=NEPB+1
        DO 50 I=1,NEPB1
50      NERB(I)=0
C       XMIT INPUT BINARY VECTOR
        IF(NEPB.LE.0)GO TO 40
        DO 30 J=1,NEPB
        DO 10 I=1,63
        ISV1=NERB(J)
        IP=I+63*(J-1)
        IM=INBA(IP)
        INERB=NERB(J)
        CALL RANERR(IM.PROB,IRN,JRN,INERB)
        INBA(IP)=IM
        NERB(J)=INERB
D       IF(ISV1.NE.NERB(J))WRITE(5,100)ICOUNT,IP
10      CONTINUE
30      CONTINUE
40      CONTINUE
C       RETURN
        ISTP=NEPB*63+1
        DO 20 I=ISTP,NBRPF
        ISV1=NERB(NEPB1)
        IM2=INBA(I)
        INERB2=NERB(NEPB1)
        CALL RANERR(IM2,PROB,IRN,JRN,INERB2)
        INBA(I)=IM2
        NERB(NEPB1)=INERB2
D       IF(ISV1.NE.NERB(NEPB1))WRITE(5,100)ICOUNT,I
20      CONTINUE
100     FORMAT(1X,'FR=',I4,2X,'ERR LC=',I3)
C       WRITE(5,809)(INBA(I),I=1,NBRPF)
        RETURN
        END
C       RANERR.FTN
        SUBROUTINE RANERR(IX,PROB,IRN,JRN,NER)
        CALL RANDU(IRN,JRN,YOR)
        IF(YOR.GE.PROB)RETURN
        IX=IEOR(IX,1)
        NER=NER+1
        RETURN
        END
```

```
      SUBROUTINE OVEVOD(LTH1,ITWID,IMARK)
C     SUBROUTINE EVODD(LTH1,XR,XI,ITWID,IMARK)
C     WE WILL TAKE A REAL SIGNAL X AND DO AN N/2 PT
C     COMPLEX FFT ON X BY BREAKING THE SIGNAL INTO ITS EVEN AND ODDPOINTS.
C     THEN THE PROPERTIES OF EVEN-
C     ODD SEPARATION WILL BE USED TO GET XR,XI THE REAL
C     AND IMAGINARY COMPONENTS OF THE FFT OF THE ORGINAL SIGNAL
C
C     ITWID JUST DETERMINES WHETHER THE FORWARD FFT IS DONE
C     OR THE INVERSE FFT. IF ITWID EQUALS ONE THE FORWARD FFT IS RETURNED.
      COMMON/SORT/DCT1(256),DCT2(256),IORDR(256),XR(512),XI(512)
C     DIMENSION  XR(1),XI(1)
      LTH=2*LTH1
      LTH2=LTH1+2
      LTH3=LTH+2
      LTHALF=LTH1/2
C     NSTAGE IS THE NUMBER OF STAGES FOR THE FFT
      NSTAGE=INT(ALOG(FLOAT(LTH1))/ALOG(2.0))
      PI=4.0*ATAN(1.0)
      PI2=2*PI/LTH
      PI3=PI2/4
C     NOW BREAK INTO EVEN AND ODD COMPONENTS
C     ALSO SCALE BY2 TO ADJUST OUTPUT
C     DO 11 J=1,LTH1
C     XR(J)=XR(2*J-1)/2.0
C     XI(J)=XR(2*J)/2.0
C11   CONTINUE
C     NOW LOAD INTO FFT AND THEN RESHUFFLE
      CALL FASTF(NSTAGE,1,1,2)
      XTEMP1=2.0*(XR(1)+XI(1))
      XR(LTHALF+1)=2.0*(XR(LTHALF+1))
      XTEMP2=2.0*(XR(1)-XI(1))
      XR(1)=XTEMP1
      XI(1)=0.0
      XI(LTHALF+1)=-2.0*(XI(LTHALF+1))
      IF(IMARK.EQ.2)GO TO 12
      XR(LTH1+1)=XTEMP2
      XR(LTH1+LTHALF+1)=XR(LTHALF+1)
      XI(LTH1+1)=0.0
      XI(LTH1+LTHALF+1)=-1.0*XI(LTHALF+1)
C
12    DO 25 K=2,LTHALF
      J=K-1
      Q1=(XI(K)+XI(LTH2-K))
      Q2=(XI(K)-XI(LTH2-K))
      Q3=(XR(K)+XR(LTH2-K))
      Q4=(XR(K)-XR(LTH2-K))
C
      QTEMP1=((SIN(PI2*J))*Q4)
      QTEMP2=((SIN(PI2*J))*Q1)
      QTEMP3=((COS(PI2*J))*Q1)
      QTEMP4=((COS(PI2*J))*Q4)
C
       XR(K)=Q3-QTEMP1+QTEMP3
       XR(LTH2-K)=Q3+QTEMP1-QTEMP3
       XI(K)=Q2-QTEMP4-QTEMP2
       XI(LTH2-K)=-1.0*(Q2+QTEMP4+QTEMP2)
       IF (IMARK.EQ.2)GO TO 25
       XR(LTH3-K)=XR(K)
       XR(LTH1+K)=XR(LTH2-K)
       XI(LTH3-K)=-1.0*XI(K)
       XI(LTH1+K)=-1.0*XI(LTH2-K)
25     CONTINUE
       IF (ITWID.EQ.1)GO TO 30
```

```
C         NOW DO COMPLEX CONJUGATE AND NORMALIZE
C         BY LTH FOR THE INVERSE FFT.
          XLTH=-1.0*LTH
          DO 26 I=1,LTH1
          XR(I)=XR(I)/LTH
          XI(I)=XI(I)/XLTH
26        CONTINUE
30        RETURN
          END
```

```
C       FASTF.FTN        14-FEB-78
        SUBROUTINE FASTF(NP,JTYPE,IS,IMARK)
C       NP = LOG NUMBER OF SAMPLES, MAX=10.
C       JTYPE=1  IF DIRECT TRANSFORM,  JTYPE=2  IF REVERSE TRANSFORM.
C       IS = 1.  IF COEFFICIENT TABLE SET-UP REQUIRED,
C       IS = 2.  IF COEFFICIENT TABLE BY-PASS DESIRED.
        COMMON/SORT/DCT1(256),DCT2(256),IORDR(256),X(512),Y(512)
        COMMON/BLOCK1/INIT1,DCT(256),LTH,NSTAGE
        DIMENSION S(256),KX(11)
        REAL*8 PIHAF
        EQUIVALENCE (KX(1),K10),(KX(2),K9),(KX(3),K8),(KX(4),K7)
        EQUIVALENCE (KX(5),K6),(KX(6),K5),(KX(7),K4),(KX(8),K3),(KX(9),K2)
        EQUIVALENCE (KX(10),K1)
        N=2**NP
         IF(IMARK.EQ.2)GO TO 1200
C       WRITE(5,934)(I,X(I),I=1,4)
C       WRITE(5,934)(I,Y(I),I=1,4)
934     FORMAT(/1X,4(1X,'FFT(',I3,')=',E15.8,2X))
1200    N4=N/4
        GO TO (1,3),IS
1       S(N4)=1.
         IF(N4-1)3,3,5
5       N8=N/8
        PI=4.0*ATAN(1.0)
        PIHAF=PI/2.
        S(N8)=DSIN(PIHAF/2.)
         IF(N8-1)3,3,6
6       N16=N/16
        S(N16)=DSIN(PIHAF/4.)
        NN16=N8+N16
        S(NN16)=DCOS(PIHAF/4.)
         IF(N16-1)3,3,7
7       M=NP-2
        LX=8
         DO 10 L=3,M
         I=N4/LX
         S(I)=DSIN(PIHAF/LX)
         IC=N4-I
         S(IC)=DCOS(PIHAF/LX)
         KMAX=LX/2-2
         LX=LX*2
         DO 10 K=1,KMAX
         KI=(2*K+1)*I
         KID=KI-I
         KIDC=N4-KID
10       S(KI)=S(I)*S(KIDC)+S(IC)*S(KID)
3        MX=1
         DO 100 M=1,NP
          INCR=MX
          NANGL=-INCR
          MX=MX*2
          JMAX=N/MX
          IDEL=2*JMAX
          DO 100 J=1,JMAX
          NANGL=NANGL+INCR
          DO 100 I=J,N,IDEL
           IJMAX=I+JMAX
           XTEM=X(I)-X(IJMAX)
           YTEM=Y(I)-Y(IJMAX)
           X(I)=X(I)+X(IJMAX)
           Y(I)=Y(I)+Y(IJMAX)
           IF(J-1)50,50,51
50         X(IJMAX)=XTEM
           Y(IJMAX)=YTEM
        GO TO 100
51      IF(NANGL-N4)53,52,54
52      GO TO (60,61),JTYPE
60      X(IJMAX)=YTEM
        Y(IJMAX)=-XTEM
        GO TO 100
61      X(IJMAX)=-YTEM
        Y(IJMAX)=XTEM
        GO TO 100
53      SN=S(NANGL)
        NCOS=N4-NANGL
        CS=S(NCOS)
        GO TO 75
54      NSIN=2*N4-NANGL
        SN=S(NSIN)
        NCOS=N4-NSIN
        CS=-S(NCOS)
75      GO TO (90,91),JTYPE
90      X(IJMAX)=XTEM*CS+YTEM*SN
        Y(IJMAX)=-XTEM*SN+YTEM*CS
        GO TO 100
91      X(IJMAX)=XTEM*CS-YTEM*SN
        Y(IJMAX)=XTEM*SN+YTEM*CS
100     CONTINUE
C       REORDER RESULTS IN NATURAL ORDER
        KX(1)=N
        DO 22 L=2,NP
22      KX(L)=KX(L-1)/2
        DO 24 L=NP,9
24      KX(L+1)=1
        IJ=1
        DO 30 J1=1,K1,1
        DO 30 J2=J1,K2,K1
        DO 30 J3=J2,K3,K2
        DO 30 J4=J3,K4,K3
        DO 30 J5=J4,K5,K4
        DO 30 J6=J5,K6,K5
        DO 30 J7=J6,K7,K6
        DO 30 J8=J7,K8,K7
        DO 30 J9=J8,K9,K8
        DO 30 JI=J9,K10,K9
        IF(IJ-JI)28,29,29
28      XTEM=X(IJ)
        X(IJ)=X(JI)
        X(JI)=XTEM
        YTEM=Y(IJ)
        Y(IJ)=Y(JI)
        Y(JI)=YTEM
29      GO TO (30,31),JTYPE
31      X(IJ)=X(IJ)/N
        Y(IJ)=Y(IJ)/N
30      IJ=IJ+1
C       K-TH SPECTRAL VALUE IS STORED IN X(K+1),Y(K+1),K=0,N-1.
        IF(IMARK.EQ.2)RETURN
C       WRITE(5,934)(I,X(I),I=1,4)
C       WRITE(5,934)(I,Y(I),I=1,4)
        RETURN
        END
```

```
C       GF2ADD.FTN
C       ADDITION OVER GF(2)
C       POLINOMIAL A(X) MUST BE ORDERED IN DESCENDING POWER SERIES
        SUBROUTINE GF2ADD(INA,NA,INB,NB,INC,NC)
        DIMENSION INA(1),INB(1),INC(1)
        NC=NA
        IF(NB.GT.NA)NC=NB
        DO 10 I=1,NC
        IC=NC+1-I
        IRA=NA+1-I
        IRB=NB+1-I
        ITA=0
        ITB=0
        IF(IRA.GT.0)ITA=INA(IRA)
        IF(IRB.GT.0)ITB=INB(IRB)
        INC(IC)=IEOR(ITA,ITB)
10      CONTINUE
        RETURN
        END
```

```
C       GF2MUL..FTN
C       MULTIPLICATION OVER GF(2)
C       NA<NF.NB<NF
        SUBROUTINE GF2MUL(INA,NA,INB,NB,INC,NC,INF,NF)
        DIMENSION IAT(17),INA(1),INB(1),INC(1),INF(1)
        NCC=NA+NB-1
C       INI VECTOR C
        DO 10 I=1,NCC
10      IAT(I)=0
C       MULTIPLY A AND B
        DO 20 I=1,NA
        DO 30 J=1,NB
        IC=I+J-1
        IT=IAND(INA(I),INB(J))
        IAT(IC)=IEOR(IAT(IC),IT)
30      CONTINUE
20      CONTINUE
        CALL GF2DIV(IAT,NCC,INF,NF,INC,NC)
        RETURN
        END
```

```
C       GF2DIV.FTN
C       FINITE FIELD DIVISION
C       DEVISOR VECTOR B IS DESTROYED IN COMPUTATION IF NOT NORMALIZWD
        SUBROUTINE GF2DIV(INA,NA,INB,NB,INC,NC)
        DIMENSION INA(1),INB(1),INC(1)
C       NORMALIZE VECTOR B
        NC=NB-1
        NBP=NB
        DO 11 I=1,NB
        IF(INB(1).EQ.1)GO TO 22
        NBP=NBP-1
        IF(NBP.LE.0)GO TO 11
        DO 33 J=1,NBP
33      INB(J)=INB(J+1)
11      CONTINUE
C       VECTOR B=0
        WRITE(5,100)
100     FORMAT(1X,'DIVISOR=0')
        RETURN
22      CONTINUE
        IF(NA.GE.NBP)GO TO 10
C       INA(I) IS THE ANSWER
        DO 20 I=1,NC
        IR=NC+1-I
        INC(IR)=0
        IF(I.GT.NA)GO TO 20
        ITA=NA+1-I
        INC(IR)=INA(ITA)
20      CONTINUE
        RETURN
10      CONTINUE
C       INI VECTOR C
C       ACTUAL NA MAY BE SMALLER THAN NBP
        DO 30 I=1,NBP
30      INC(I)=INA(I)
        NAP=NBP
111     CONTINUE
C       CHECK C(1)=1
        IF(INC(1).EQ.0)GO TO 222
C       START DIVISION
        DO 50 I=1,NBP
50      INC(I)=IEOR(INC(I),INB(I))
222     CONTINUE
        NAP=NAP+1
        IF(NAP.GT.NA)GO TO 333
C       SHIFT ONE BIT LEFT
        DO 60 I=1,NBP-1
60      INC(I)=INC(I+1)
        INC(NBP)=INA(NAP)
        GO TO 111
333     CONTINUE
C       INSERT 0 IF NBP NOT EQUAL TO NB
        IF(NBP.NE.NB)GO TO 777
        DO 555 I=1,NC
555     INC(I)=INC(I+1)
        RETURN
777     CONTINUE
        DO 773 I=1,NC
        IT=0
        IR=NC+1-I
        IBC=NBP+1-I
        IF(I.GT.NBP)GO TO 773
        IT=INC(IBC)
773     INC(IR)=IT
```

```
RETURN
END
```

```
      SUBROUTINE TAPE3(IQ)
      IMPLICIT INTEGER(A-Z)
      COMMON/MTAPE0/NIN(256).NOUT(256)
      COMMON/MTAPE1/NSKIP,IST,NTOTI,NTUPS,NTOTO
      COMMON/MTAPE2/NEND,NEPR,NFILE.NINS,NOUTS
      COMMON/MTAPE3/NBF(1324),NBUF(1324)
      COMMON/MTAPE4/LST.IBEG
      DIMENSION ICARD(64)
     GO TO (700,900,900,999,1000.1001,1002,1003).IQ
C
C    INPUT DATA
700     IST=IST+NTUPS
        IF(1324.GE.IST) GO TO 200
  100 IST=IST-1024
      NSKIP=NSKIP+1
200   KOVER=LST+NTOTI-1325
        IF(KOVER.GT.0) GO TO 305
        DO 5000 I=1,NTOTI
5000    NIN(I)=NBF(LST+I-1)
      LST=LST+NTUPS
        RETURN
  305 IPEP=LST-1024
        DO 5001 I=1,300
5001    NBF(IPEP+I-1)=NBF(LST+I-1)
      LST=IPEP
        IF(NINS.EQ.0) GO TO 2100
C       DISK INPUT
        DO 5010 I=1,16
        READ(2,END=2001,ERR=6000)(ICARD(J),J=1,64)
        K=64*(I-1)+300
        DO 5010 J=1,64
5010    NBF(K+J)=ICARD(J)
        IF(NERR.NE.0) GO TO 6000
        GO TO 200
2001    NEND=1
        GO TO 200
2100    CALL TIN(NBF(301),NEND,NERR)
      GO TO 200
C
  900 CONTINUE
C     OUTPUT DATA
        DO 5005 I=1,NTOTO
5005    NBUF(IBEG+I-1)=NOUT(I)
        IBEG=IBEG+NTOTO
        KON=1025-IBEG
        IF(KON.GT.0) GO TO 90
        IF(NOUTS.EQ.0) GO TO 2002
        DO 6010 I=1,16
        K=64*(I-1)
        DO 6011 J=1,64
6011    ICARD(J)=NBUF(K+J)
6010    WRITE(3)(ICARD(J),J=1,64)
        GO TO 2003
2002    CALL TOUT(NBUF,NEPR)
2003    LRESID=-KON
        DO 5006 I=1,LRESID
5006    NBUF(I)=NBUF(1024+I)
      IBEG=LRESID+1
90      RETURN
C
C     INITIALIZE
900     IF((NINS+NOUTS).LE.1) CALL ATTACH
        IF(NINS.EQ.0) CALL RWND0
        IF(NOUTS.EQ.0) CALL RWND1
      IF(NFILE.EQ.0) GO TO 995
      IF(NINS.EQ.0) CALL FSRCH(NFILE,NERR)
      IF(NERR.NE.0) GO TO 6000
913   DO 912 J=1,NSKIP
      IF(NINS.EQ.0) GO TO 3000
C     DISK INPUT
      DO 5011 I=1,16
      READ(2,END=3001,ERR=6000)(ICARD(JJ),JJ=1,64)
      K=64*(I-1)+300
      DO 5011 JJ=1,64
5011  NBF(K+JJ)=ICARD(JJ)
      GO TO 912
3001  NEND=1
      GO TO 912
3000  CALL TIN(NBF(301),NEND,NERR)
      IF(NERR.NE.0) GO TO 6000
  912 CONTINUE
      GO TO 995
  999 CONTINUE
      IBEG=1
      CALL EOFSH(NERR)
      RETURN
  995 CONTINUE
      IBEG=1
      LST=IST+300
      IST=IST-NTUPS
      RETURN
C     END OF FILE
1000  IF(NOUTS.EQ.0) GO TO 2010
      CALL CLOSE(3)
      GO TO 2011
2010  CALL EOFW(NERR)
2011  IBEG=1
      RETURN
1001  NEND=0
      IF(NINS.EQ.0) GO TO 4020
      REWIND 2
      GO TO 4011
4020  CALL RWND0
      NERR=0
      CALL FSRCH(NFILE,NERR)
      IF(NERR.NE.0) GO TO 6000
4011  DO 950 J=1,NSKIP
      IF(NINS.EQ.0) GO TO 4000
C     DISK INPUT
      DO 5012 I=1,16
      READ(2,END=4001,ERR=6000)(ICARD(J1),J1=1,64)
      K=64*(I-1)+300
      DO 5012 J2=1,64
5012  NBF(K+J2)=ICARD(J2)
      GO TO 950
4001  NEND=1
      GO TO 4002
4000  CALL TIN(NBF(301),NEND,NERR)
      IF(NERR.NE.0) GO TO 6000
4002  IF(NEND.NE.0) GO TO 2000
950   CONTINUE
      LST=IST+300
      IST=IST-NTUPS
      RETURN
2000  NERR=16384
      RETURN
1002  IF(NINS.EQ.1)CALL CLOSE(2)
      NEND=0
```

```
1003    CALL INFO
        RETURN
5000    TYPE 6001
6001    FORMAT(1X,'INPUT FILE ERROR'/)
        NERR=1
        RETURN
      END
C       PROGRAM FSIO.FTN TO MOVE MAG TAPES
C       AND WRITE SPEECH FOR REAL-TIME I/O USING QIO
C
        SUBROUTINE ATTACH
        IMPLICIT INTEGER(A-Z)
        COMMON/MTAPE2/NEND,NERR,NFILE,NINS,NOUTS
        COMMON/MTAPE5/MASK,ISW(2),IOATT,IOSUC,IEALN,IORWD,
     1  IOWLB,IEVER,IOSPF,IEEOF,IOEOF,IORLB,MT0(6),MT1(6),DSW
        DATA IOATT,IOSUC,IEALN/O001400,1,-34/
        DATA IORWD,IOWLB,IEVER,IOSPF,IORLB/O2400,O400,-4,O2440,O1000/
        DATA IOSPF,IEEOF,IOEOF/O2440,-10,O3000/
        DATA MASK/O377/
        DATA MT0/0,2048,0,0,0,0/
        DATA MT1/0,2048,0,0,0,0/
        IF(NINS.NE.0) GO TO 1
        CALL ASNLUN(2,'MT',0,DSW)
        IF(DSW.EQ.1)GO TO 10
11      WRITE(5,100)
100     FORMAT(1X,'MT0: ATTACH UNSUCCESSFUL'/)
        NERR=1
        RETURN
10      CALL WTQIO(IOATT,2,1,0,ISW,0,DSW)
        IF(IOSUC.EQ.IAND(MASK,ISW(1)))GO TO 1
        IF(IAND(IEALN,MASK).NE.IAND(MASK,ISW(1)))GO TO 11
1       IF(NOUTS.NE.0) GO TO 2
        CALL ASNLUN(3,'MT',1,DSW)
        IF(DSW.EQ.1) GO TO 20
12      WRITE(5,101)
101     FORMAT(1X,'MT1: ATTACH UNSUCCESSFUL'/)
        NERR=1
        RETURN
20      CALL WTQIO(IOATT,3,1,0,ISW,0,DSW)
        IF(IOSUC.EQ.IAND(MASK,ISW(1)))GO TO 2
        IF(IAND(IEALN,MASK).NE.IAND(MASK,ISW(1)))GO TO 12
2       RETURN
        END
C
        SUBROUTINE RWND0
        IMPLICIT INTEGER(A-Z)
        COMMON/MTAPE2/NEND,NERR,NFILE,NINS,NOUTS
        COMMON/MTAPE5/MASK,ISW(2),IOATT,IOSUC,IEALN,IORWD,
     1  IOWLB,IEVER,IOSPF,IEEOF,IOEOF,IORLB,MT0(6),MT1(6),DSW
        CALL WTQIO(IORWD,2,1,0,ISW,0,DSW)
        IF(IOSUC.EQ.IAND(MASK,ISW(1)))GO TO 1
        WRITE(5,902)
902     FORMAT(1X,'MT0: BUSY'/)
        NERR=1
1       RETURN
        END
C
        SUBROUTINE RWND1
        IMPLICIT INTEGER(A-Z)
        COMMON/MTAPE2/NEND,NERR,NFILE,NINS,NOUTS
        COMMON/MTAPE5/MASK,ISW(2),IOATT,IOSUC,IEALN,IORWD,
     1  IOWLB,IEVER,IOSPF,IEEOF,IOEOF,IORLB,MT0(6),MT1(6),DSW
        CALL WTQIO(IORWD,3,1,0,ISW,0,DSW)
        IF(IOSUC.EQ.IAND(MASK,ISW(1)))GO TO 1
        WRITE(5,902)
      NERR=1
902   FORMAT(1X,'MT1: BUSY'/)
1     RETURN
      END
C
      SUBROUTINE TIN(BUF,NEND,NERR)
      IMPLICIT INTEGER(A-Z)
      COMMON/MTAPE5/MASK,ISW(2),IOATT,IOSUC,IEALN,IORWD,
     1 IOWLB,IEVER,IOSPF,IEEOF,IOEOF,IORLB,MT0(6),MT1(6),DSW
      NEND=0
      NERR=0
      CALL GETADR(MT0,BUF)
      CALL WTQIO(IORLB,2,1,0,ISW,MT0,DSW)
      IF(IOSUC.EQ.IAND(MASK,ISW(1)))GO TO 1
      IF(IAND(IEEOF,MASK).EQ.IAND(MASK,ISW(1)))NEND=1
      IF(IAND(IEVER,MASK).EQ.IAND(MASK,ISW(1)))NERR=1
1     RETURN
      END
C
      SUBROUTINE TOUT(NBUF,NERR)
      IMPLICIT INTEGER(A-Z)
      COMMON/MTAPE5/MASK,ISW(2),IOATT,IOSUC,IEALN,IORWD,
     1 IOWLB,IEVER,IOSPF,IEEOF,IOEOF,IORLB,MT0(6),MT1(6),DSW
      NERR=0
      CALL GETADR(MT1,NBUF)
      CALL WTQIO(IOWLB,3,1,0,ISW,MT1,DSW)
      IF(IOSUC.EQ.IAND(MASK,ISW(1)))GO TO 1
      IF(IAND(IEVER,MASK).EQ.IAND(MASK,ISW(1)))NERR=1
1     RETURN
      END
C
      SUBROUTINE FSRCH(NFILE,NERR)
      IMPLICIT INTEGER (A-Z)
      COMMON/MTAPE3/NBF(1324),NBUF(1324)
      COMMON/MTAPE5/MASK,ISW(2),IOATT,IOSUC,IEALN,IORWD,
     1 IOWLB,IEVER,IOSPF,IEEOF,IOEOF,IORLB,MT0(6),MT1(6),DSW
      NERR=0
      FILE=NFILE-1
      IF(FILE.LE.0) RETURN
      DO 1 I=1,FILE
      CALL GETADR(MT0,NBF(301))
      CALL WTQIO(IORLB,2,1,0,ISW,MT0,DSW)
      IF(IOSUC.EQ.IAND(MASK,ISW(1)))GOTO 2
      WRITE(5,100)NFILE
100   FORMAT(1X,'FILE',I4, ' NOT FOUND'/)
      NERR=1
      RETURN
2     MT0(1)=1
1     CALL WTQIO(IOSPF,2,1,0,ISW,MT0,DSW)
      RETURN
      END
C
      SUBROUTINE EOFSH(NERR)
      IMPLICIT INTEGER(A-Z)
      COMMON/MTAPE3/NBF(1324),NBUF(1324)
      COMMON/MTAPE5/MASK,ISW(2),IOATT,IOSUC,IEALN,IORWD,
     1 IOWLB,IEVER,IOSPF,IEEOF,IOEOF,IORLB,MT0(6),MT1(6),DSW
      NERR=0
1     CALL GETADR(MT1(1),NBUF(1))
      CALL WTQIO(IORLB,3,1,0,ISW,MT1,DSW)
      IF(IOSUC.EQ.IAND(MASK,ISW(1)))GO TO 1
      IF(IAND(IEEOF,MASK).EQ.IAND(MASK,ISW(1)))GO TO 2
      NERR=1
      WRITE(5,1000)ISW(1)
1000  FORMAT(1X,'FILE SEARCH ERROR' O9/)
```

```
      RETURN
2     CALL GETADR(MT1(1),NBUF(1))
      CALL WTQIO(IORLB,3,1,0,ISW,MT1,DSW)
      IF(IOSUC.EQ.IAND(MASK,ISW(1)))GO TO 1
      IF(IAND(IEEOF,MASK).EQ.IAND(MASK,ISW(1))) GO TO 3
      NERR=1
      RETURN
3     MT1(1)=-1
      CALL WTQIO(IOSPF,3,1,0,ISW,MT1,DSW)
      RETURN
      END
C
      SUBROUTINE EOFW(NERR)
      IMPLICIT INTEGER(A-Z)
      COMMON/MTAPES/MASK,ISW(2),IOATT,IOSUC,IEALN,IORWD,
     1 IOWLB,IEVER,IOSPF,IEEOF,IOEOF,IORLB,MT0(6),MT1(6),DSW
      NERR=0
      DO 1 I=1,2
1     CALL WTQIO(IOEOF,3,1,0,ISW)
      IF(IOSUC.EQ.IAND(MASK,ISW(1)))GO TO 2
      NERR=1
      RETURN
2     MT1(1)=-1
      CALL WTQIO(IOSPF,3,1,0,ISW,MT1,DSW)
      IF(IOSUC.EQ.IAND(MASK,ISW(1)))RETURN
      NERR=1
      RETURN
      END
C     SUBROUTINE INFO.FTN
C
      SUBROUTINE INFO
      IMPLICIT INTEGER(A-Z)
      COMMON/MTAPE0/NIN(256),NOUT(256)
      COMMON/MTAPE1/NSKIP,IST,NTOTI,NTUPS,NTOTO
      COMMON/MTAPE2/NEND,NERR,NFILE,NINS,NOUTS
      COMMON/MTAPE3/NBF(1324),NBUF(1324)
      LOGICAL*1 Y,ANS
      LOGICAL*1 EXTIN,EXTOUT,APPEND
      DIMENSION EXTIN(3),EXTOUT(3)
      DATA EXTIN/'I','N','P'/
      DATA EXTOUT/'O','U','T'/
      DATA Y/'Y'/
      DATA NEND,NERR,NFILE/0,0,0/
      DATA NSKIP,IST/1,1/
C
C     GET I/O INFORMATION  FOR SPEECH HANDLER
C
      APPEND=.FALSE.
      TYPE 1
1     FORMAT(1H$,'IS THE INPUT ON MAG. TAPE? ')
      READ(5,2)ANS
2     FORMAT(A1)
      IF(ANS.NE.Y)NINS=1
      NINS=1
      TYPE 3
3     FORMAT(1H$'IS THE OUTPUT GOING TO MAG TAPE? ')
      READ (5,2) ANS
      IF(ANS.NE.Y) NOUTS=1
      NOUTS=1
      IF(NOUTS.NE.0) GO TO 5
      TYPE 4
4     FORMAT(1H$,'APPEND DATA? ')
      READ (5,2) ANS
      IF(ANS.EQ.Y) APPEND=.TRUE.
5     IF(NOUTS.EQ.0) GO TO 151
```

```
C       RSX11  SUPPORTED FILE
        TYPE 150
150     FORMAT(1H$,'OUTPUT FILE NAME= ')
        CALL FILEN(3,EXTOUT)
C
C       BEGINNINE OF INPUT
C
151     IF(NINS.NE.0) GO TO 155
        TYPE 100
100     FORMAT(1H$'MT FILE NO.=(I3) ')
        READ(5,101)NFILE
101     FORMAT(I3)
        GO TO 14
155     TYPE 13
13      FORMAT(1H$,'INPUT FILE NAME= ')
        CALL FILEN(2,EXTIN)
        NFILE=1
14      CALL TAPE3(3)
        IF(NERR.NE.0) RETURN
        IF(APPEND)CALL TAPE3(4)
        RETURN
        END
C
        SUBROUTINE FILEN(UNIT,EXT)
C
C       THIS SUBROUTINE ACCEPTS THE NAME OF THE INPUT OR OUTPUT FILE
C       FORM THE TTY DEVICE 5
C       DEFAULT DEVICE
C       UNLESS SPECIFIED IN INPUT STRING
C       UNIT=UNIT NUMBER
C       EXT = LOGICAL*1 BUFFER OF EXTENSION
C
        IMPLICIT INTEGER(A-Z)
        LOGICAL*1 INSTR,DOT,BLNK,EXT
        DIMENSION INSTR(40)
        DIMENSION EXT(3)
        DATA BLNK,DOT/' ','.'/
C
C       INPUT FILE
152     READ (5,99)(INSTR(I),I=1,40)
99      FORMAT(40A1)
C       CHECK FOR END OF LINE
        DO 1600 I=40,1,-1
        J=I
1600    IF(INSTR(I).NE.BLNK)GO TO 1601
        TYPE 151
151     FORMAT(1H$,'>')
        GO TO 152
1601    DO 1602 I=1,J
        IF(INSTR(I).NE.BLNK) GO TO 1602
C
C
C       BLANK DISCOVERED-COLLAPSE LINE BY ONE AND DECREASE CHARACTER COUNT
C
        DO 1603 K=I,J-1
1603    INSTR(K)=INSTR(K+1)
        INSTR(J)=BLNK
        J=J-1
        GO TO 1601
1602    CONTINUE
        DO 103 I=1,J
103     IF(INSTR(I).EQ.DOT) GO TO 25
        INSTR(J+1)=DOT
        INSTR(J+2)=EXT(1)
        INSTR(J+3)=EXT(2)
        INSTR(J+4)=EXT(3)
        J=J+4
35      CALL SCAN(INSTR.J)
        CALL ASSIGN(UNIT,INSTR.J)
        RETURN
        END
C
        SUBROUTINE SCAN(BUF,LTH)
        IMPLICIT INTEGER(A-Z)
        LOGICAL*1 BUF,DEVICE
        DIMENSION BUF(1),DEVICE(4)
        DATA DEVICE/'S','Y'.'0',':'/
        DO 1 I=1,LTH
1       IF(BUF(I).EQ.DEVICE(4))RETURN
        LTH=LTH+4
        DO 2 I=LTH,5,-1
2       BUF(I)=BUF(I-4)
        DO 3 I=1,4
3       BUF(I)=DEVICE(I)
        RETURN
        END
```